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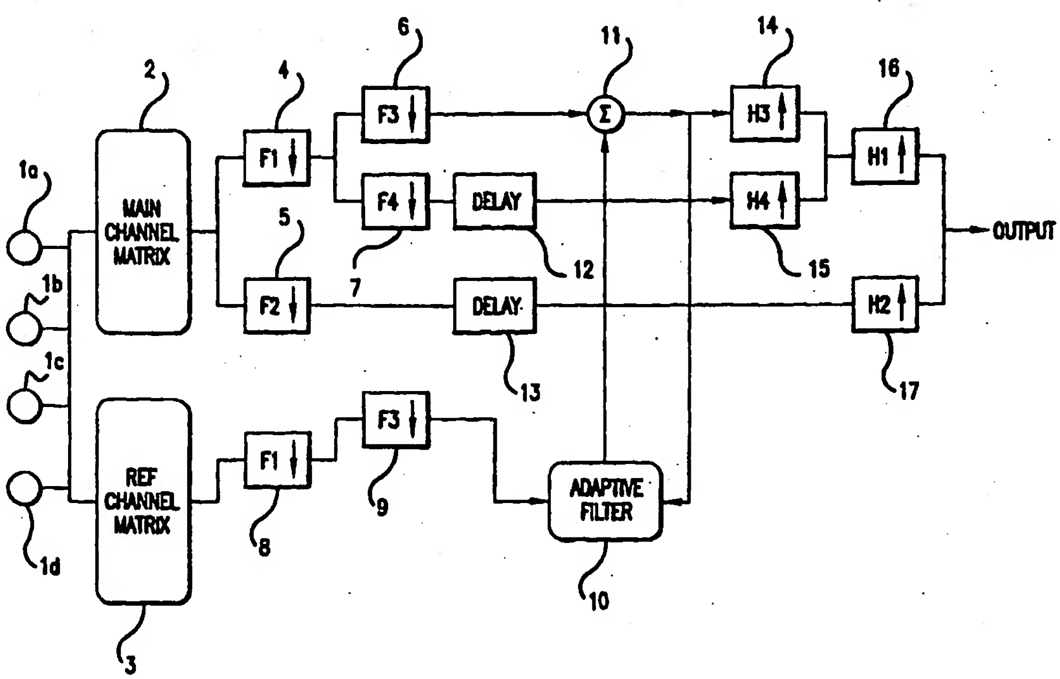

(57) Abstract

A dual-processing interference cancelling system and method for processing a broadband input in a computationally fficient manner. Dual processing divides the input into higher and lower frequency bands and applies adaptive filter processing to the lower frequency band while applying non-adaptive filter processing to the higher frequency band. Various embodiments are shown including those based on sub-bands, broadband processing with band-limited adaptation, and broadband processing with an external main-channel generator.

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DUAL-PROCESSING INTERFERENCE CANCELLING SYSTEM AND METHOD

BACKGROUND OF THE INVENTION

The present invention relates generally to signal processing, and more specifically to an interference cancelling system and method using a combination of adaptive and non-adaptive filter processing. A system using such a combination of adaptive and non-adaptive filter processing is referred to herein as a dual-processing system.

Interference cancelling systems have a wide range of applications such as directional microphones and hearing aids. An interference cancelling system amplifies a target signal originating from a target source (information source) while suppressing interfering signals ("interferences") originating from interference or noise sources.

Interference cancelling systems using adaptive filters are well-known in the art. An adaptive filter is a filter which can change its characteristics by changing its filter coefficients. The interference cancelling system may be a non-directional system having one or more sensors measuring the signal received from the target to generate a main channel, which has a target signal component and an interference component. The system may include one or more other sensors for measuring the interferences to generate one or more reference channels. The adaptive filter uses the reference channels to cancel the interference component present in the main channel.

Alternatively, the system may be a directional system, well-known in the art, which amplifies a target signal originating from a target source at a particular direction relative to the system and suppresses interferences

originating from interference sources at all other directions. In such a directional system, the target signal and the interferences may be detected by an array of spatially distributed sensors forming what is called a beamformer.

A beamformer is a form of spatial filter, itself well-known in the art, which takes inputs from an array of spatially distributed sensors and combines them in such a way that it either enhances or suppresses signals coming from certain directions relative to signals from other directions. Thus it can change the direction of receiving sensitivity without physically moving the sensor array. The inputs are combined for this purpose based on filter coefficients as discussed below.

In non-adaptive beamforming, the filter coefficients of a beamformer are predetermined such that the beamformer can exhibit maximum sensitivity or minimum sensitivity (null) in a predetermined direction. Since the coefficient values are fixed in time, a non-adaptive beamformer cannot dynamically place nulls in the directions of strong interferences existing at particular times as the environment changes.

In adaptive beamforming, in contrast, the spatial filter coefficients of a beamformer are continually updated so that directional sensitivity can be dynamically changed depending on the changing locations of a target source and interference sources. For more details on beamforming, see Van Veen & Buckley, Beamforming: A Versatile Approach to Spatial Filtering, IEEE ASSP Magazine, April 1988, pp. 4-24.

An adaptive beamformer can be implemented for example by using tapped delay lines, forming a finite-impulse-response (FIR) filter having time-varying coefficients which are directly changed as the locations of interference sources change.

Alternatively, the adaptive beamformer can be implemented using an adaptive filter (dealing with temporal signals rather than spatial signals). The adaptiv beamformer uses fixed-coefficient tapped delay lines, called

a main-channel matrix, to obtain a signal received from the direction of a target and other fixed-coefficient tapped delay lines, called a reference-channel matrix, to obtain interferences received from all other directions. An adaptive filter is used to generate cancelling signals resembling the interferences changing in direction. In this manner, instead of directly changing the coefficients of the tapped delay lines, the implementation achieves the same effect by changing the characteristics of the adaptive filter. The adaptive filter generally subtracts the cancelling signals from the main channel and adjusts the filter weights to minimize the mean-square values of the output. When the filter weights settle, the cancelling signals closely track the interferences so that the output has substantially reduced interference.

For some applications, it is important to be able to process a broadband input, that is, one having a relatively large bandwidth. For example, in hearing applications, speech intelligibility is critical to performance. It is well known that the higher frequency portion of the speech spectrum carries much of the information required for speech intelligibility. For applications such as hearing aids or directional microphones for voice activation systems, good intelligibility requires at least 6 Khz of bandwidth. In fact, professional audio systems will not tolerate a bandwidth of less than 12 Khz.

This bandwidth requirement imposes a severe computational burden on the interference cancelling system using adaptive filter processing. Adaptive filter processing is inherently intensive in computation. It involves performing filter operations to produce an output and further updating filter weights based on the output. All these operations must be performed for each new sample.

In order to extend the operation of an adaptive filter in the discrete time domain from any bandwidth to a broader bandwidth, the sampling rate should be increased to maintain comparable quality. According to the well-known sampling

theorem, a sampling rate of at least twice the maximum frequency of an incoming analog signal is required in order to represent the signal completely in the discrete time domain. The increased sampling rate increases the number of operations to be performed per unit time.

Increasing the sampling rate alone is not, however, enough to handle the broader bandwidth. An adaptive filter acts on later samples by observing earlier samples within a given period, as feedback. How well the adaptive filter can react depends on how long the filter can observe the earlier samples. This time period is called an effective time delay through an adaptive filter. The delay is proportional to the number of filter stages, each storing a filter coefficient, divided by the sampling frequency. If the sampling frequency is increased, the number of filter stages should be increased in order to maintain the same effective time delay. The increased number of filter stages also increases the number of operations that must be performed per unit time.

The combination of increasing sampling rate and increasing the number of required filter stages sharply increases the number of operations to be performed by a processor. Thus a simple extension of adaptive filter processing to a broader bandwidth places a disproportionately large computational burden on the system and hence is not desirable.

The simple extension of adaptive filter processing presents another problem for an interference cancelling system using adaptive filter processing. Adaptive interference cancelling systems suffer from signal leakage. The system works well when the reference channel is uncorrelated to the main channel. However, in practice, the reference channel contains some signals correlated to the main channel due to signal leakage from the main channel itself. Adaptive filter processing may then partly cancel the target signal as well the inferences. The signal leakage is more likely to occur at higher frequencies for the following reason.

The reference-channel matrix produces reference channels by creating a null in the target direction (by suppressing signals from the target direction). In order to suppress the signals from the target direction effectively, the null should be as deep as possible in the target direction. The null should also be wide enough to provide some tolerance to those signals slightly off the target direction. It turns out that the null is much wider at lower frequencies than at higher frequencies. Therefore, any mismatch in the sensor array would impact the effectiveness of the null much less at lower frequencies than at higher frequencies. In other words, the system is much more sensitive to a mismatch at higher frequencies than at lower frequencies.

Therefore, there exists a need for an improved interference cancelling system that can process an input of given bandwidth without significantly increasing computational requirements and without the drawbacks of adaptive filter processing at higher frequencies. We note that the invention is applicable to a system of any bandwidth; no minimum bandwidth for its application is intended since it can provide advantages in terms of processing efficiencies or capabilities for any bandwidth.

SUMMARY OF THE INVENTION

Accordingly, it is an object of the present invention to provide an interference cancelling system capable of processing a broadband input without disproportionately increasing the computational burden.

Another object of the invention is to provide an interference cancelling system which can avoid the problems ordinarily encountered at higher frequencies with adaptive filter processing.

These and other objects are achieved in accordance with the present invention by dividing an input spectrum into lower and upper sub-bands and applying adaptive filter processing to the lower sub-band while applying non-adaptive filter processing to the upper sub-band. This dual

processing is based on the recognition that the performance of adaptive filter processing becomes worse at higher frequencies. Since non-adaptive filter processing is much lower in computational burden, the overall result is better, performing broadband processing with a significantly lower computational burden.

In a preferred embodiment, a main channel and reference channels are obtained using nonadaptive filter processing. The main channel is then split into lower and upper sub-bands. The reference channels are also split in the same way, but only the lower sub-bands are kept while the upper sub-bands are discarded. An adaptive filter uses the lower sub-band of the main channel and the lower sub-bands of the reference channels to generate cancelling signals which are then subtracted from the lower sub-band of the main channel to produce a lower sub-band output. The lower sub-band output is combined with the upper sub-band of the main channel to reconstruct the broadband output.

In another preferred embodiment, a broadband main channel and broadband reference channels are obtained using non-adaptive filter processing. The broadband main channel or the broadband reference channels are not divided into sub-bands. Instead, the broadband reference channels are low-pass filtered to drive an adaptive filter in the low frequency band to obtain low-frequency cancelling signals. The low-frequency cancelling signals are converted to broadband cancelling signals by up-sampling so that they can be subtracted from the broadband main channel over its entire bandwidth.

In yet another preferred embodiment, an external main-channel generator, such as a commercially available hi-fidelity directional microphone, is used in place of a main matrix to obtain a broadband main channel by taking advantage of the broadband capability of existing hi-fidelity microphones. A low-frequency reference matrix generates low-frequency references, which, in turn, drives an adaptive filter to generate low-frequency cancelling signals. The

low-frequency cancelling signals are translated to a broadband cancelling signals by up-sampling so that they can be subtracted from the broadband main channel.

The above-stated objects are preferably achieved in accordance with the present invention using methods which can, as will be apparent to those knowledgeable in this field, readily be implemented in a program controlling a commercially available digital signal processor or a general-purpose microprocessor.

BRIEF DESCRIPTION OF THE DRAWINGS

The objects, features, and advantages of the present invention will be more readily apparent from the following detailed description of the invention in which:

FIG. 1 is a block diagram of a system using sub-band processing;

FIG. 2 is a block diagram of a system using broadband processing with frequency-limited adaptation;

FIG. 3 is a block diagram of a system using broadband processing with an external main-channel generator;

FIGS. 4A-4D are a flow chart depicting the operation of a program that may be used to implement a method using sub-band processing;

FIGS. 5A-5C are a flow chart depicting the operation of a program that may be used to implement a method using broad-band processing with frequency-limited adaptation; and

FIGS. 6A-6C are a flow chart depicting the operation of a program that may be used to implement a method using broad-band processing with an external main-channel generator.

DETAILED DESCRIPTION OF THE INVENTION

A. System Implementation

1. Sub-band Processing

FIG. 1 shows one preferred embodiment of the present invention using sub-bands where an adaptive filter driven from the sub-bands rather than the entire bandwidth of the input signal. Sub-bands result from partitioning a broader

band in any manner as long as the subbands can be combined together so that the broader band can be reconstructed without distortions. One may use a so-called "perfect reconstruction structure" as known in the art to split the broadband into sub-bands and to combine the sub-bands together substantially without distortion. For details on perfect reconstruction structures, see P.P. Vaidyanathan, Quadrature Mirror Filter Banks, M-Band Extensions and Perfect-Reconstruction Techniques, IEEE ASSP Magazine, pp. 4-20, July 1987.

In the preferred embodiment, a broader band is partitioned into sub-bands, using several partitioning steps successively through intermediate bands. Broadband inputs from an array of sensors, 1a-1d, are sampled at an appropriate sampling frequency and entered into a main-channel matrix 2 and a reference-channel matrix 3. The main-channel matrix generates a main channel, a signal received in the main looking direction of the sensor array, which contain a target signal component and an interference component. Alternatively, the main channel may be provided by an external main-channel generator such as a shot-gun microphone, a parabolic microphone, or a dipole microphone.

F1, 4, and F2, 5 are splitters which first split the main channel into two intermediate bands, followed by down-sampling by two. Down-sampling is a well-known procedure in digital signal processing. Down-sampling by two, for example, is a process of sub-sampling by taking every other data point. Down-sampling is indicated by a downward arrow in the figure. Splitters F3, 6 and F4, 7 further split the lower intermediate band into two sub-bands followed by down-sampling by two.

In an example using a 16 Khz input signal, the result is a 0-4 Khz lower sub-band with 1/4 of the input sampling rate, a 4-8 Khz upper sub-band with 1/4 of the input sampling rate, and another upper 8-16 Khz intermediate band with 1/2 of the input sampling rate.

The reference channels are processed in the same way by filters F1, 8, and F2, 9, to provide only the lower sub-band with 1/4 of the input sampling rate, while the other sub-bands are discarded.

The lower sub-bands of the reference channels are fed into an adaptive filter 10, which generates cancelling signals approximating interferences present the main channel. A subtracter 11 subtracts the cancelling signals from the lower sub-band of the main channel to generate an output in the lower sub-band. The output is fed back to the adaptive filter for updating the filter weights. The adaptive filter processing and the subtraction is performed at the lower sampling rate appropriate for the lower sub-band. At the same time the other upper bands of the main channel are delayed by delay units, 12 and 13, each by an appropriate time, to compensate for various delays caused by the different processing each sub-band is going through, and to synchronize them with the other sub-bands. The delay units may be implemented by a series of registers or a programmable delay. The output from the subtracter is combined with the other two sub-bands of the main channel through the reconstruction filters H1-H4, 14-17, to reconstruct a broadband output. H1-H4 may be designed such that they together with F1-F4 provide a theoretically perfect reconstruction without any distortions.

Reconstructors H3 and H4 combine the lower and upper sub-bands into a low intermediate band, followed by an interpolation by two. An interpolation is a well-known procedure in digital signal processing. Interpolation by two, for example, is an up-sampling process increasing the number of samples by taking every other data point and interpolating them to fill as samples in between. Up-sampling is indicated by an upward arrow in the figure. The reconstructors H1, 16 and H2, 17 further combine the two intermediate bands into a broadband.

In the preferred embodiment described, non-adaptive filter processing is performed in th upper sub-band of 4-16

Khz. Adaptive filter processing is performed in the lower sub-band of 0-4 Khz where most of interferences are located. Since there is little computation overhead involved in the non-adaptive filter processing, the use of non-adaptive filter processing in the upper sub-band can reduce the computational burden significantly. The result is superior performance without an expensive increase in the required hardware.

2. Broadband Processing with Band-Limited Adaptation

FIG. 2 shows another preferred embodiment using broadband processing with band-limited adaptation. Instead of using sub-band cancelling signals which act on a sub-band main channel, the embodiment uses broadband cancelling signals which act on a broadband main channel. But, since adaptive filter processing is done in a low-frequency domain, the resulting cancelling signals are converted to a broadband signal so that it can be subtracted from the broadband main channel.

As before, broadband inputs from an array of sensors, 21a-21d, are sampled at an appropriate sampling frequency and entered into a main-channel matrix 22 and a reference-channel matrix 23. The main-channel matrix generates a main channel, a signal received in the main-looking direction, which has a target signal component and an interference component. The reference-channel matrix generates reference channels representing interferences received from all other directions. A low-pass filter 25 filters the reference channels and down-samples them to provide low-frequency signals to an adaptive filter 26.

The adaptive filter 26 acts on these low-frequency signals to generate low-frequency cancelling signals which estimate a low-frequency portion of the interference component of the main channel. The low-frequency cancelling signals are converted to broadband signals by an interpolator 28 so that they can be subtracted from the main channel by a subtracter 29 to produce a broadband output.

The broadband output is low-pass filtered and down-sampled by a filter 24 to provide a low-frequency feedback signal to the adaptive filter 26. In the mean time, the main channel is delayed by a delay unit 27 to synchronize it with the cancelling signals from the adaptive filter 26.

3. Broadband Processing with an External Main-Channel Generator

FIG. 3 shows yet another preferred embodiment similar to the previous embodiment except that an external main-channel generator is used instead of a main-channel matrix to obtain a broadband main channel. This embodiment is useful when it is desired to take advantage of the broadband capabilities of commercially available hi-fi microphones.

A broadband input is obtained by using an external main-channel generator, such as a shotgun microphone 43, a parabolic dish 44, or a dipole microphone. The broadband input is sampled through a high fidelity A-to-D converter 45. The sampling rate should preferably be high enough to maintain the broad bandwidth and the audio quality of the external main-channel generator.

A reference-channel matrix 42 is used to obtain low-frequency reference channels representing interferences in the low-frequency domain. Since adaptive filter processing is done in the low-frequency domain, the reference-channel matrix does not need a broadband capability.

A subtracter 50 is used to subtract cancelling signals estimating interferences from the broadband input. The broadband output is filtered by a low-pass filter 46 which also performs down-sampling. The low-pass filtered output and the low-frequency reference channels are provided to an adaptive filter 47. The adaptive filter acts on these low frequency signals to generate low-frequency cancelling signals. In the meantime, the broadband input is delayed by a delay unit 48 so that it can be synchronized with the cancelling signals from the adaptive filter 47. The delay unit may be implemented by a series of registers or by a

programmable delay. The low-frequency cancelling signals are converted to broadband cancelling signals by an interpolator 49 so that they can be subtracted from the broadband main channel to produce the broadband output.

It is noted that the adaptive filter used in the present invention is not limited to a particular kind of adaptive filter. For example, one can practice the present invention using the invention disclosed in applicant's commonly assigned and copending U.S. patent application Serial No. 08/672,899, filed June 27, 1996, entitled 'System and Method for Adaptive Interference Cancelling,' by inventor Joseph Marash and its corresponding PCT application WO 97/50186, published December 31, 1997. Both applications are incorporated by reference herein in their entirety.

Specifically, the adaptive filter may include weight constraining means for truncating updated filter weight values to predetermined threshold values when each of the updated filter weight value exceeds the corresponding threshold value. The adaptive filter may further include inhibiting means for estimating the power of the main channel and the power of the reference channels and for generating an inhibit signal to the weight updating means based on normalized power difference between the main channel and the reference channels.

The weight constraining means may include a frequency-selective weight-control unit, which includes a Fast Fourier Transform (FFT) unit for receiving adaptive filter weights and performing the FFT of the filer weights to obtain frequency representation values, a set of frequency bins for storing the frequency representation values divided into a set of frequency bands, a set of truncating units for comparing the frequency representation values with a threshold assigned to each bin and for truncating the values if they exceed the threshold, a set of storage cells for temporarily storing the truncated values, and an Inverse Fast

Fourier Transform (IFFT) unit for converting them back to the adaptive filter weights.

B. Software Implementation

The invention described herein may be implemented using a commercially available digital signal processor (DSP) such as Analog Device's 2100 Series or any other general purpose microprocessor. For more information on Analog Device 2100 Series, see Analog Device, ADSP-2100 Family User's Manual, 3rd Ed., 1995.

1. Sub-Band Processing

FIGS. 4A-4D are a flow chart depicting the operation of a program in accordance with the first preferred embodiment of the present invention using sub-band processing.

Upon starting at step 100, the program initializes registers and pointers as well as buffers (steps 110-120). When a sampling unit sends an interrupt (step 131) that samples are ready, the program reads the sample values (step 130), and stores them in memory (step 140).

The program retrieves the input values (step 151) and main-channel matrix coefficients (step 152) to generate a main channel by filtering the inputs values using the coefficients (step 150), and then stores the result in memory (step 160).

The program retrieves the input values (step 171) and reference-channel matrix coefficients (step 172) to generate a reference channel by filtering the input values using the coefficients (step 170), and then store the result (step 180). Steps 170 and 180 are repeated to generate all other reference channels (step 190).

The program retrieves the main channel (step 201) and the F1 filter coefficients (step 202) to generate an lower intermediate band with 1/2 of the sampling rate appropriate for the whole main channel by filtering the main channel with the coefficients and down-sampling the filtered output (step

210), and then stores the result (step 220). Similarly, the F2 filter coefficients are used to generate a upper intermediate band with 1/2 of the sampling rate (step 240). The F3 and F3 filter coefficients are used to further generate a lower sub-band with 1/4 of the sampling rate (step 260) and a upper sub-band with 1/4 of the sampling rate (step 280).

The program retrieves one of the reference channels (step 291) and the F1 filter coefficients (step 292) to generate an intermediate band with 1/2 of the sampling rate by filtering the reference channel with the coefficients and down-sampling the filtered output (step 290), and then stores the result (step 300). Similarly, the F2 filter coefficients are used to generate a lower sub-band with 1/4 of the sampling rate (step 320). Steps 290-320 are repeated for all the other reference channels (step 330).

The program retrieves the reference channels (step 341) and the main channel (step 342) to generate cancelling signal using an adaptive beamforming process routine (step 340). The program subtracts the cancelling signals from the main channel to cancel the interference component in the main channel (step 350).

The program then interpolates the output from the adaptive beamforming process routine (step 360) and filtering the output with the H3 filter coefficients (step 361) to obtain an up-sampled version (step 370). The program also interpolates the main channel in the lower band (step 380) and filters it with the H4 filter coefficients (step 381) to obtain an up-sampled version (step 390). The program combines the up-sampled versions to obtain a lower intermediate main channel (step 400).

The program interpolates the lower intermediate main channel (step 410) and filters it with the H1 filter coefficients (step 420) to obtain an up-sampled version (step 420). The program also interpolates the upper intermediate main channel (step 430) and filters it with the H2 filter coefficients (step 431) to obtain an up-sampled version (step

440). The program combines the up-sampled versions to obtain a broadband output (step 450).

2. Broadband Processing with Frequency-Limited Adaptation

FIGS. 5A-5C are a flow chart depicting the operation of a program in accordance with the second preferred embodiment of the present invention using broadband processing with frequency-limited adaptation.

Upon starting at step 500, the program initializes registers and pointers as well as buffers (steps 510-520). When a sampling unit sends an interrupt (step 531) that the samples are ready, the program reads the sample values (step 530), and stores them in memory (step 540).

The program retrieves the broadband sample values (step 551) and the main-channel matrix coefficients (step 552) to generate a broadband main channel by filtering the broadband sample values with the coefficients (step 550), and then stores the result in memory (step 560).

The program retrieves the broadband samples (step 571) and reference-channel matrix coefficients (step 572) to generate a broadband reference channel by filtering the samples using the coefficients (step 570), and then stores the result (step 580). Steps 570 and 580 are repeated to generate all the other reference channels (step 590).

The program retrieves the reference channels (step 601) which are down-sampled (step 602), the main channel (step 603) which is also down-sampled to the low sampling rate (step 604), and the low-frequency output (step 605) to generate a low-frequency cancelling signal (step 600) using an adaptive beamforming process routine. The program updates the adaptive filter weights (step 610) and interpolates the low-frequency cancelling signal to generate a broadband cancelling signal (step 620). Steps 610-620 are repeated for all the other reference channels (step 630).

The program subtracts the cancelling signals from the main channel to cancel the interference component in the main channel (step 640).

The program filters and interpolates the broadband output (step 650) so that the low-frequency output can fed back to update the adaptive filter weights.

3. Broadband Processing with an External Main-Channel Generator

FIGS. 6A-6C are a flow chart depicting the operation of a program in accordance with the third preferred embodiment of the present invention using broadband processing with an external main-channel generator.

Upon starting at step 700, the program initializes registers and pointers as well as buffers (steps 710-720). When a sampling unit sends an interrupt (step 731) that samples are ready, the program reads the sample values (step 730), and stores them in memory (step 740).

The program then reads a broadband input from the external main-channel generator (step 750), and stores it as a main channel (step 760).

The program retrieves the low-frequency input (step 771) and reference-channel matrix coefficients (step 772) to generate a reference channel by multiplying the two (step 770), and then stores the result (step 780). Steps 770 and 780 are repeated to generate all the other reference channels (step 790).

The program retrieves the low-frequency reference channels (step 801), the main channel (step 802) which is down-sampled (step 803), and a low-frequency output (step 604) to generate low-frequency cancelling signals (step 600) using an adaptive beamforming process routine. The program updates the adaptive filter weights (step 810) and interpolates the low-frequency cancelling signal to generate the broadband cancelling signal (step 820). Steps 810-820 are repeated for all the other reference channels (step 830).

The program subtracts the broadband cancelling signals from the broadband main channel to generate the broadband output with substantially reduced interferences (step 840).

The program low-pass filters and interpolates the broadband output (step 850) so that the low-frequency output can fed back to update the adaptive filter weights.

While the invention has been described with reference to several preferred embodiments, it is not intended to be limited to those embodiments. It will be appreciated by those of ordinary skill in the art that many modifications may be made to the structure and form of the described embodiments without departing from the spirit and scope of the invention, which is defined and limited only in the following claims. As but one example, one of the reference channels may be obtained by measuring the vibration of an interference source using an accelerometer instead of using a microphone. The disclosed invention may also be used for processing radar signals from a phased-array antenna, or any other phenomena producing oscillatory waves detectable by any means whatsoever.

WHAT IS CLAIMED IS:

1. A dual-processing interference cancelling system for processing an input containing a target signal originating from a target source and interferences originating from interference sources and for producing an output representing the target signal with substantially reduced interferences, comprising:

a main-channel generator capable of receiving signals from such input and for generating therefrom a main channel representing signals received from the target source and having a target signal component and an interference component;

a first splitter, connected to the main-channel generator, for splitting the main channel into lower and upper sub-bands, wherein the lower and upper sub-bands together form the entire main channel;

a reference-channel generator capable of receiving signals from such input and for generating therefrom one or more reference channels representing signals received from the interference sources;

a second splitter, connected to the reference-channel generator, for splitting said one or more reference channels into lower and upper sub-bands, wherein the lower and upper sub-bands for each reference channel together form the entire reference channel;

an adaptive filter, having filter weights, connected to receive the lower sub-bands of said one or more reference channels for generating one or more cancelling signals approximating an interference component of the lower sub-band of the main channel;

a subtracter, connected to the first splitter and the adaptive filter, for generating an output by subtracting said one or more cancelling signals from the lower sub-band of the main channel;

the adaptive filter also being connected to receive the output from the subtracter and said system including filter-weight-updating means for determining updated filter weight

values for the adaptive filter such that the differences between the lower sub-band of the main channel and the cancelling signals are substantially minimized; and

a reconstructor, connected to the subtracter and to the first splitter, for reconstructing a broadband output by combining the upper sub-band of the main channel and the output from the subtracter.

2. The system of claim 1, further comprising a first set of one or more sensors for receiving signals from a target source and a second set of one or more sensors for receiving interferences.

3. The system of claim 2, wherein said the sensors in the first set and the second set are microphones.

4. The system of claim 2, wherein one or more sensors of the second set are accelerometers for sensing vibration of a surrounding structure.

5. The system of claim 1, further comprising one or more sensors for receiving signals from a target source and also for receiving signals from interferences sources.

6. The system of claim 1, wherein said main-channel generator is a main-channel matrix which generates a main channel from an array of sensors, the main channel representing signals received in the direction of the target.

7. The system of claim 1, wherein said reference-channel generator is a reference-channel matrix which generates reference channels from an array of sensors, the reference channels representing signals received in the directions other than the direction of the target.

8. The system of claim 1, wherein said adaptive filter comprises a finite-impulse-response filter for generating said one or more cancelling signals.

9. The system of claim 1, wherein said adaptive filter comprises an infinite-impulse-response filter for generating said one or more cancelling signals.

10. The system of claim 1, wherein said filter-weight-updating means uses the least-mean-square algorithm where the mean-square values of the differences between the lower sub-

band of the main channel and said one or more cancelling signals are substantially minimized.

11. A dual-processing interference cancelling system for processing an input containing a target signal as well as interferences and for producing an output representing the target signal with substantially reduced interferences, comprising:

a main-channel matrix capable of receiving signals from such input and for generating therefrom a main channel representing signals received from the direction of the target source and having a target signal component and an interference component;

a first splitter, connected to the main-channel matrix, for splitting the main channel into lower and upper sub-bands, wherein the lower and upper sub-bands together form the entire main channel;

a reference-channel matrix capable of receiving signals from signal sources and for generating therefrom reference channels representing signals received from the directions other than the direction of the target source;

a second splitter, connected to the reference-channel matrix, for splitting the reference channels into lower and upper sub-bands, wherein the lower and upper sub-bands of each reference channel together form the entire reference channel;

an adaptive filter, having filter weights, connected to receive the lower sub-band reference channels for generating cancelling signals approximating an interference component of the lower sub-band main channel;

a subtracter, connected to the main channel splitter and the adaptive filter, for generating a sub-band output by subtracting the cancelling signals from the lower sub-band main channel;

the adaptive filter also being connected to receive the output from the subtracter and said system including filter-weight-updating means for determining updated filter weight

values for the adaptive filter such that the differences between the lower sub-band of the main channel and the cancelling signals are substantially minimized; and

a reconstructor, connected to the subtracter and to the main-channel matrix, for reconstructing a broadband output by combining the upper sub-band of the main channel and the sub-band output from the subtracter.

12. The system of claim 11, wherein said adaptive filter comprises a finite-impulse-response filter for generating the cancelling signals.

13. The system of claim 11, wherein said adaptive filter comprises an infinite-impulse-response filter for generating the cancelling signals.

14. The system of claim 11, wherein said filter-weight-updating means uses the least-mean-square algorithm.

15. The system of claim 11, further comprising:

a sensor array of spatially distributed sensors, each for receiving the target signal and interferences;

a sampling unit, connected to receive signals from the sensor array, for converting such signals to digital form and for sending them to the main-channel matrix and the reference-channel matrix; and

an output digital-to-analog converter, connected to the subtracter, for converting said broadband output to analog form.

16. The system of claim 15, wherein the sensors are microphones.

17. The system of claim 15 wherein the first splitter comprises a down-sampler.

18. The system of claim 15 wherein the second splitter comprises a down-sampler.

19. The system of claim 15, wherein the reconstructor comprises an interpolator.

20. A dual-processing interference cancelling system for processing an input containing a target signal originating from a target source and interferences

originating from interference source and for producing an output representing the target signal with substantially reduced interferences, comprising:

a main-channel generator capable of receiving signals from such input and for generating therefrom a broadband main channel representing signals received from a target source and having a target signal component and an interference component;

a reference-channel generator capable of receiving signals from such input and for generating one or more broadband reference channels representing signals received from interference sources;

a low-pass filter, connected to the reference-channel generator, for filtering said one or more broadband reference channels into one or more low-frequency reference channels;

an adaptive filter, having filter weights, connected to receive said one or more low-frequency reference channels for generating one or more low-frequency cancelling signals approximating low-frequency interferences present in the broadband main channel;

an interpolator, connected to the adaptive filter, for interpolating said one or more low-frequency cancelling signals to one or more broadband cancelling signals;

a subtracter, connected the main-channel generator and the interpolator, for generating a broadband output by subtracting said one or more broadband cancelling signals from the broadband main channel;

a second filter for filtering the broadband output to produce a low-frequency output; and

the adaptive filter also being connected to receive the low-frequency output and said system including filter-weight-updating means for determining updated filter weight values for the adaptive filter such that the differences between the low-frequency output and said one or more low-frequency cancelling signals are substantially minimized.

21. The system of claim 20, further comprising a first set of one or more sensors for receiving signals from a

target and a second set of one or more sensors for receiving interferences.

22. The system of claim 21, wherein the sensors in the first set and the second set are microphones.

23. The system of claim 21, wherein one or more sensors of the second set are accelerometers for sensing vibration of a surrounding structure.

24. The system of claim 20, further comprising one or more sensors for receiving signals from a target and for receiving interferences.

25. The system of claim 20, wherein said main-channel generator is a main-channel matrix which generates a main channel from an array of sensors, the main channel representing signals received in the direction of the target source.

26. The system of claim 20, wherein said reference-channel generator is a reference-channel matrix which generates reference channels from an array of sensors, the reference channels representing signals received in the directions other than the direction of the target source.

27. The system of claim 20, wherein said adaptive filter comprises a finite-impulse-response filter for generating said one or more low-frequency cancelling signals.

28. The system of claim 20, wherein said adaptive filter comprises an infinite-impulse-response filter for generating said one or more low-frequency cancelling signals.

29. The system of claim 20, wherein said filter-weight-updating means uses the least-mean-square algorithm where the mean-square values of the differences between the low-frequency output and said one or more low-frequency cancelling signals are substantially minimized.

30. A dual-processing interference cancelling system for processing an input representing an input containing a target signal originating from a target source as well as interferences originating from interferences and for

producing an output representing the target signal with substantially reduced interferences, comprising:

a main-channel matrix capable of receiving signals from such input and for generating therefrom a broadband main channel representing signals received from the direction of the target and having a target signal component and an interference component;

a reference-channel matrix capable of receiving signals from such input and for generating therefrom broadband reference channels representing interferences received from the directions other than that of the target;

a first low-pass filter, connected to the reference-channel matrix, for filtering the broadband reference channels into low-frequency reference channels;

an adaptive filter, having filter weights, connected to receive the low-frequency reference channels for generating a low-frequency cancelling signals approximating the interference component of the low-frequency main channel;

an interpolator, connected to the adaptive filter, for interpolating the low-frequency cancelling signals to broadband cancelling signals;

a subtracter, connected the main-channel matrix and the interpolator, for generating a broadband output by subtracting the broadband cancelling signals from the broadband main channel;

an second low-pass filter, connected to the subtracter, for filtering the broadband output to get a low-frequency output; and

the adaptive filter also being connected to receive the low-frequency output and said system including filter-weight-updating means for determining updated filter weight values for the adaptive filter such that the differences between the low-frequency output and the low-frequency cancelling signals are substantially minimized.

31. The system of claim 30, wherein said adaptive filter comprises a finite-impulse-response filter for generating said one or more low-frequency cancelling signals.

32. The system of claim 30, wherein said adaptive filter comprises an infinite-impulse-response filter for generating said one or more low-frequency cancelling signals.

33. The system of claim 30, wherein said filter-weight-updating means uses the least-mean-square algorithm.

34. The system of claim 30, further comprising:

a sensor array of spatially distributed sensors, each for receiving the target signal and interferences;

a sampling unit, connected to receive signals from the sensor array, for converting such signals to digital form and for sending them to the main-channel matrix and the reference-channel matrix; and

an output digital-to-analog converter, connected to the subtracter, for converting said broadband output to analog form.

35. The system of claim 34, wherein the sensors are microphones.

36. A dual-processing interference cancelling system for processing an input containing a target signal originating from a target source and interferences originating from interference sources and for producing an output representing a target signal with substantially reduced interferences, comprising:

an external main-channel generator for capable of receiving signals from one or more signal sources and generating therefrom a broadband main channel representing signals received from the target source and having a target signal component and an interference component;

a low-frequency reference-channel generator capable of receiving signals from such input and for generating therefrom one or more low-frequency reference channels representing low-frequency interferences;

an adaptive filter, having filter weights, connected to the low-frequency reference-channel generator, for generating one or more low-frequency cancelling signals approximating

low-frequency interferences present in the interference component of the broadband main channel;

an interpolator, connected to the adaptive filter, for interpolating the low-frequency cancelling signals to broadband cancelling signals;

a subtracter, connected to the external main-channel generator and the interpolator, for generating a broadband output by subtracting the broadband cancelling signals from the broadband main channel;

a low-pass filter, connected to the subtracter, for filtering the broadband output to get a low-frequency output; and

the adaptive filter also being connected to receive the low-frequency output and said system including filter-weight-updating means for determining updated filter weight values for the adaptive filter such that the differences between the low-frequency output and said one or more low-frequency cancelling signals are substantially minimized.

37. The system of claim 36, wherein the external main-channel generator comprises a sensor for receiving signals from the target.

38. The system of claim 37, wherein the sensor is a shot-gun microphone.

39. The system of claim 37, wherein the sensor is a parabolic microphone.

40. The system of claim 36, wherein said reference-channel generator is a reference-channel matrix generating reference channels from an array of sensors, each reference channel representing an interference from a direction other than the direction of the target signal.

41. The system of claim 36, wherein said adaptive filter comprises a finite-impulse-response filter for generating said one or more low-frequency cancelling signals.

42. The system of claim 36, wherein said adaptive filter comprises an infinite-impulse-response filter for generating one or more low-frequency cancelling signals.

43. The system of claim 36, wherein said filter-weight-updating means uses the least-mean-square algorithm where the mean-square value of the differences between the lower sub-band of the main channel and said one or more low-frequency cancelling signals is substantially minimized.

44. A dual-processing interference cancelling system for processing an input containing a target signal originating from a target source as well as interferences originating from interference sources and for producing an output representing a target signal with substantially reduced interferences, comprising:

an external main-channel generator capable of receiving signals from one or more signal sources and for generating therefrom a broadband main channel having a target signal component and an interference component;

a low-frequency reference-channel matrix capable of receiving signals from such input and for generating therefrom low-frequency reference channels representing low-frequency signals received in the directions other than the direction of the target source;

an adaptive filter, having filter weights, connected to the reference-channel matrix, for generating low-frequency cancelling signals approximating the interference component of the main channel;

an interpolator, connected to the adaptive filter, for interpolating the low-frequency cancelling signals to broadband cancelling signals;

a subtracter, connected to the external main-channel generator and the interpolator, for generating a broadband output by subtracting the broadband cancelling signals from the broadband main channel;

a low-pass filter, connected to the subtracter, for filtering the broadband output to get a low-frequency output; and

the adaptive filter also being connected to receive the low-frequency output and said system including filter-weight-

updating means for determining updated filter weight values for the adaptive filter such that the differences between the low-frequency output and the low-frequency cancelling signals are substantially minimized.

45. The system of claim 44, wherein said adaptive filter comprises a finite-impulse-response filter for generating the low-frequency cancelling signals.

46. The system of claim 44, wherein said adaptive filter comprises an infinite-impulse-response filter for generating the low-frequency cancelling signals.

47. The system of claim 44, wherein the filter-weight-updating means uses the least-mean-square algorithm.

48. The system of claim 44, further comprising:

a sensor array of spatially distributed sensors, each for receiving the target signal and interferences;

a sampling unit, connected to receive signals from the sensor array, for converting such signals to digital form and for sending them to the low-frequency reference-channel matrix; and

an output digital-to-analog converter, connected to the subtracter, for converting the broadband output to analog form.

49. The system of claim 48, wherein the sensors are microphones.

50. A method for processing an input containing a target signal originating from a target source and interferences originating from interference sources and for producing an output representing the target signal with substantially reduced interferences, comprising the steps of:

generating a broadband main channel from such input, the broadband main channel representing signals received from the target source and having a target signal component and an interference component;

splitting the broadband main channel into lower and higher sub-band main channels;

generating one or more broadband reference channels representing signals received from interference sources;

splitting said one or more broadband reference channels into one or more lower and upper sub-band reference channels;

generating one or more lower sub-band cancelling signal approximating the interference component in the main channel by filtering said one or more lower sub-band reference channels using an adaptive filter;

generating a lower sub-band output by subtracting said one or more lower sub-band cancelling signals from the lower sub-band main channel;

reconstructing a broadband output by combining the lower sub-band output and the upper sub-band main channel; and

adaptively adjusting filter weights of the adaptive filter so that the differences between the lower sub-band main channel and said one or more lower sub-band cancelling signals are substantially minimized.

51. The method of claim 50, wherein the step of generating said one or more lower sub-band cancelling signals uses a finite-impulse-response filter.

52. The method of claim 50, wherein the step of generating said one or more lower sub-band cancelling signals uses an infinite-impulse-response filter.

53. The method of claim 50, wherein the step of adaptively updating filter weights uses the least-mean-square algorithm.

54. The method of claim 50, further comprising the steps of:

generating input signals from a sensor array of spatially distributed sensors, each for receiving the target signal and interferences;

sampling the inputs signal and converting them to digital form before the step of generating a broadband main channel and the step of generating one or more broadband reference channels; and

converting the broadband output to analog form after the combining step.

55. The method of claim 54, wherein the sensors are microphones.

56. A method for processing an input containing a target signal originating from a target source and interferences originating from interference sources and for producing an output representing the target signal with substantially reduced interferences, comprising the steps of:

generating a broadband main channel, the broadband main channel representing signals received from the target source and having a target signal component and an interference component;

generating one or more broadband reference channels representing signals received from interference sources;

filtering said one or more broadband reference channels to get one or more low-frequency reference channels;

generating one or more low-frequency cancelling signals approximating the interference component in the main channel by filtering said one or more low-frequency reference channels using an adaptive filter;

interpolating said one or more low-frequency cancelling signals to generate one or more broadband cancelling signals;

generating a broadband output by subtracting said one or more broadband cancelling signals from the broadband main channel;

filtering the broadband output to generate a low-frequency output; and

adaptively updating filter weights of the adaptive filter so that the differences between the low-frequency output and said one or more low-frequency cancelling signals are substantially minimized;

57. The method of claim 56, wherein the step of generating one or more low-frequency cancelling signals uses a finite-impulse-response filter.

58. The method of claim 56, wherein the step of generating one or more low-frequency cancelling signal uses an infinite-impulse-response filter.

59. The method of claim 56, wherein the step of adaptively updating filter weights uses the least-mean-square algorithm.

60. The method of claim 56, further comprising the steps of:

generating input signals from a sensor array of spatially distributed sensors, each for receiving the target signal and the interferences;

sampling the input signals and converting them to digital form before the step of generating a broadband main channel and the step of generating broadband reference channels; and

converting the broadband output to analog form after the step of generating a broadband output.

61. The method of claim 60, wherein the sensors are microphones.

62. A method for processing an input containing a target signal originating from a target source and interferences originating from interference sources and for producing an output representing the target signal with substantially reduced interferences, comprising the steps of:

generating a broadband main channel using an external main-channel generator, the broadband main channel representing a signal received from the target source and having a target signal component and an interference component;

generating one or more low-frequency reference channels from such input, said one or more low-frequency reference channels representing signals received from the interference sources;

generating one or more low-frequency cancelling signals approximating the interference component in the main channel by filtering said one or more low-frequency reference channels using an adaptive filter;

interpolating said one or more low-frequency cancelling signals to generate a broadband cancelling signal;

generating a broadband output by subtracting the broadband cancelling signal from the broadband main channel;

filtering the broadband output to generate a low-frequency output; and

adaptively updating filter weights of the adaptive filter so that the differences between the low-frequency output and said one or more low-frequency cancelling signals are substantially minimized;

63. The method of claim 62, wherein the step of generating one or more low-frequency cancelling signals uses a finite-impulse-response filter.

64. The method of claim 62, wherein the step of generating one or more low-frequency cancelling signals uses an infinite-impulse-response filter.

65. The method of claim 62, wherein the step of adaptively updating filter weights uses the least-mean-square algorithm where the mean-square value of the differences between the-low frequency output and said one or more low-frequency cancelling signals are substantially minimized.

66. The method of claim 62, further comprising the steps of:

generating input signals from a sensor array of spatially distributed sensors, each sensor for receiving the target signal and interferences;

sampling the input signals and converting them to digital form before the step of generating the low-frequency reference channel; and

converting the broadband output to analog form after the step of generating a broadband output.

67. The method of claim 66, wherein the sensors are microphones.

68. The system of claim 1, wherein the main channel generator is an external main-channel generator.

69. The system of claim 68, wherein the external main-channel generator comprises a shot-gun microphone.

70. The system of claim 68, wherein the external main-channel generator comprises a dipole microphone.

71. The method of claim 50, wherein the step of obtaining a broadband main channel uses an external main-channel generator.

72. The method of claim 71, wherein the external main-channel generator uses a shot-gun microphone.

73. The method of claim 71, wherein the external main-channel generator uses a dipole microphone.

74. The system of claim 1, wherein the adaptive filter further comprises:

weight constraining means for truncating said new filter weight values to predetermined threshold values when each of the new filter weight values exceeds the corresponding threshold value.

75. The system of claim 74, wherein the adaptive filter further comprises:

inhibiting means, connected to receive signals from the first splitter and the second splitter, for estimating the power of the lower sub-bands of the main channel and the power of the lower sub-bands of said one or more reference channels and for generating an inhibit signal to said filter-weight-updating means when a normalized power difference between the lower sub-bands of the main channel and the lower sub-bands of said one or more one reference channel is positive.

76. The system of claim 1, wherein adaptive filter further comprises:

the weight constraining means for converting the updated filter weight values to frequency representation values, truncating the frequency representation values to predetermined threshold values, and converting them back to adaptive filter weights.

77. The system of claim 76, wherein the weight constraining means comprises:

a Fast Fourier Transform unit for generating frequency representation values of the updated filter weight values;

a set of frequency bins, each frequency bin for storing the frequency representation values for a frequency band assigned to each frequency bin;

a set of truncating means, each connected to the corresponding frequency bin, for truncating the frequency representation values stored in each frequency bin to a predetermined threshold value if the frequency representation values exceed the threshold value associated with each frequency bin; and

an Inverse Fast Fourier Transform unit, connected to the set of truncating means, for converting values from the set of truncating means back to adaptive filter weights.

78. The system of claim 11, wherein the adaptive filter further comprises:

weight constraining means for truncating said new filter weight values to predetermined threshold values when each of the new filter weight values exceeds the corresponding threshold value.

79. The system of claim 78, wherein the adaptive filter further comprises:

inhibiting means, connected to receive signals from the first splitter and the second splitter, for estimating the power of the lower sub-bands of the main channel and the power of the lower sub-bands of said one or more reference channels and for generating an inhibit signal to said weight updating means when a normalized power difference between the lower sub-bands of the main channel and the lower sub-bands of said one or more one reference channel is positive.

80. The system of claim 11, wherein adaptive filter further comprises:

weight constraining means for converting the updated filter weight values to frequency representation values, truncating the frequency representation values to

predetermined threshold values, and converting them back to adaptive filter weights.

81. The system of claim 80, wherein the weight constraining means comprises:

a Fast Fourier Transform unit for generating frequency representation values of the updated filter weight values;

a set of frequency bins, each frequency bin for storing the frequency representation values for a frequency band assigned to each frequency bin;

a set of truncating means, each connected to the corresponding frequency bin, for truncating the frequency representation values stored in each frequency bin to a predetermined threshold value if the frequency representation values exceed the threshold value associated with each frequency bin; and

an Inverse Fast Fourier Transform unit, connected to the set of truncating means, for converting values from the set of truncating means back to adaptive filter weights.

82. The system of claim 20, wherein the adaptive filter further comprises:

weight constraining means for truncating said updated filter weight values to predetermined threshold values when each of the updated filter weight values exceeds the corresponding threshold value.

83. The system of claim 82, wherein the adaptive filter further comprises:

inhibiting means, connected to receive signals from the first low-pass filter and the second low-pass filter, for estimating the power of the low-frequency output and the power of the low-frequency reference channels and for generating an inhibit signal to said filter-weight-updating means when a normalized power difference between the low-frequency output and the low-frequency reference channels is positive.

84. The system of claim 20, wherein adaptive filter further comprises:

weight constraining means for converting the updated filter weight values to frequency representation values, truncating the frequency representation values to predetermined threshold values, and converting them back to adaptive filter weights.

85. The system of claim 84, wherein the weight constraining means comprises:

a Fast Fourier Transform unit for generating frequency representation values of the updated filter weight values;

a set of frequency bins, each frequency bin for storing the frequency representation values for a frequency band assigned to each frequency bin;

a set of truncating means, each connected to the corresponding frequency bin, for truncating the frequency representation values stored in each frequency bin to a predetermined threshold value if the frequency representation values exceed the threshold value associated with each frequency bin; and

an Inverse Fast Fourier Transform unit, connected to the set of truncating means, for converting values from the set of truncating means back to adaptive filter weights.

86. The system of claim 30, wherein the adaptive filter further comprises:

weight constraining means for truncating said updated filter weight values to predetermined threshold values when each of the updated filter weight values exceeds the corresponding threshold value.

87. The system of claim 86, wherein the adaptive filter further comprises:

inhibiting means, connected to receive signals from the first low-pass filter and the second low-pass filter, for estimating the power of the low-frequency output and the power of the low-frequency reference channels and for generating an inhibit signal to said filter-weight-updating means when a normalized power difference between the low-

frequency output and the low-frequency reference channels is positive.

88. The system of claim 30, wherein adaptive filter further comprises:

weight constraining means for converting the updated filter weight values to frequency representation values, truncating the frequency representation values to predetermined threshold values, and converting them back to adaptive filter weights.

89. The system of claim 88, wherein the weight constraining means comprises:

a Fast Fourier Transform unit for generating frequency representation values of the updated filter weight values;

a set of frequency bins, each frequency bin for storing the frequency representation values for a frequency band assigned to each frequency bin;

a set of truncating means, each connected to the corresponding frequency bin, for truncating the frequency representation values stored in each frequency bin to a predetermined threshold value if the frequency representation values exceed the threshold value associated with each frequency bin; and

an Inverse Fast Fourier Transform unit, connected to the set of truncating means, for converting values from the set of truncating means back to adaptive filter weights.

90. The system of claim 36, wherein the adaptive filter further comprises:

weight constraining means for truncating said updated filter weight values to predetermined threshold values when each of the updated filter weight values exceeds the corresponding threshold value.

91. The system of claim 90, wherein the adaptive filter further comprises:

inhibiting means, connected to receive signals from the low-pass filter and the low-frequency reference-channel generator, for estimating the power of the broadband main

channel and the power of the broadband reference channels and for generating an inhibit signal to said filter-weight-updating means when a normaliz d power difference between the low-frequency output and the low-frequency reference channels is positive.

92. The system of claim 36, wherein adaptive filter further comprises:

weight constraining means for converting the updated filter weight values to frequency representation values, truncating the frequency representation values to predetermined threshold values, and converting them back to adaptive filter weights.

93. The system of claim 92, wherein the weight constraining means comprises:

a Fast Fourier Transform unit for generating frequency representation values of the updated filter weight values;

a set of frequency bins, each frequency bin for storing the frequency representation values for a frequency band assigned to each frequency bin;

a set of truncating means, each connected to the corresponding frequency bin, for truncating the frequency representation values stored in each frequency bin to a predetermined threshold value if the frequency representation values exceed the threshold value associated with each frequency bin; and

an Inverse Fast Fourier Transform unit, connected to the set of truncating means, for converting values from the set of truncating means back to adaptive filter weights.

94. The system of claim 44, wherein the adaptive filter further comprises:

weight constraining means for truncating said updated filter weight values to predetermined threshold values when each of the updated filter weight values exceeds the corresponding threshold value.

95. The system of claim 94, wherein the adaptive filter further comprises:

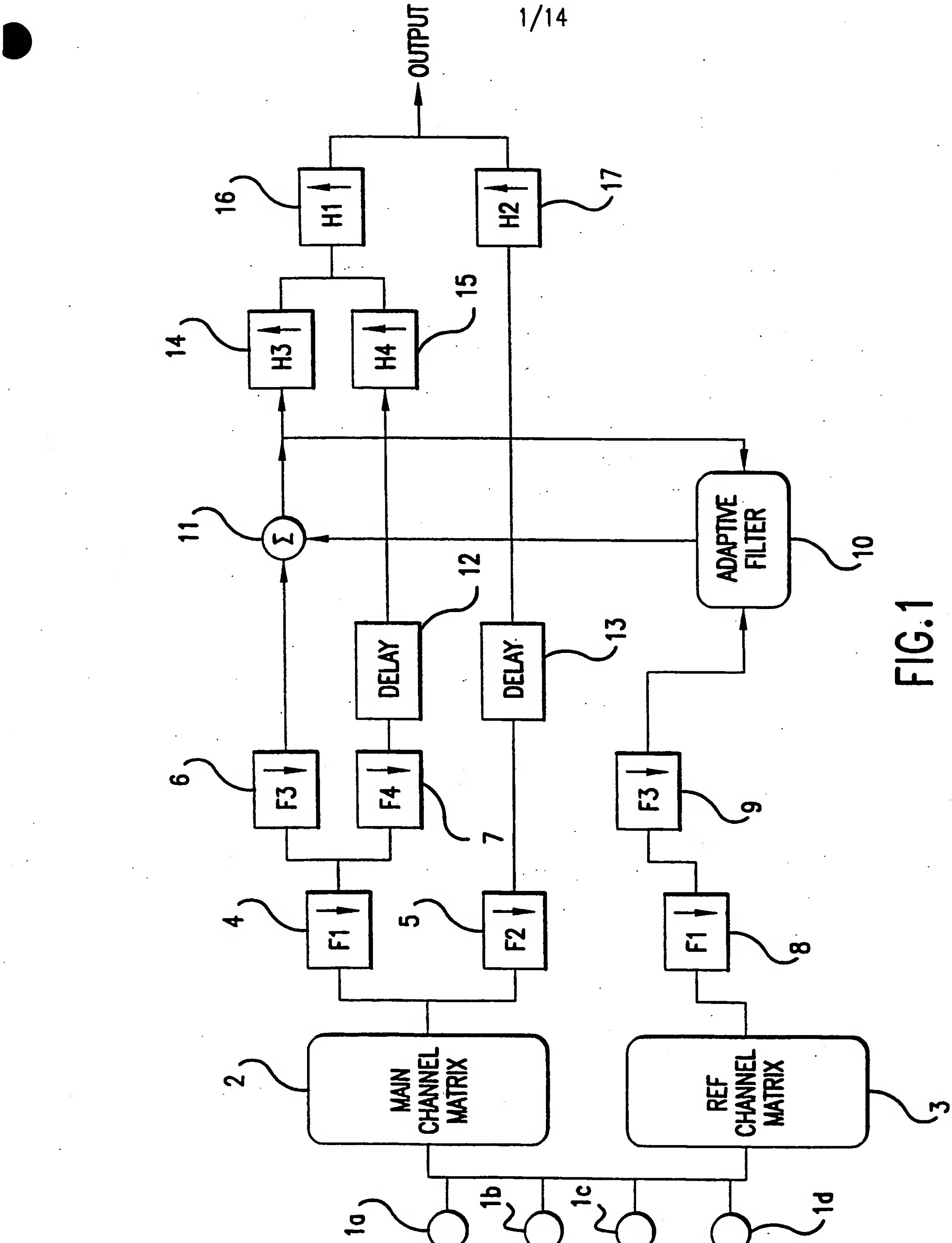
OUTPUT
H1
H2
H3
H4
16
17
14
15
Σ
11
ADAPTIVE FILTER
10
DELAY
12
DELAY
13
F3
6
F4
7
F3
9
F1
4
F2
5
F1
8
MAIN CHANNEL MATRIX
2
REF CHANNEL MATRIX
3
1a
1b
1c
1d

FIG.1

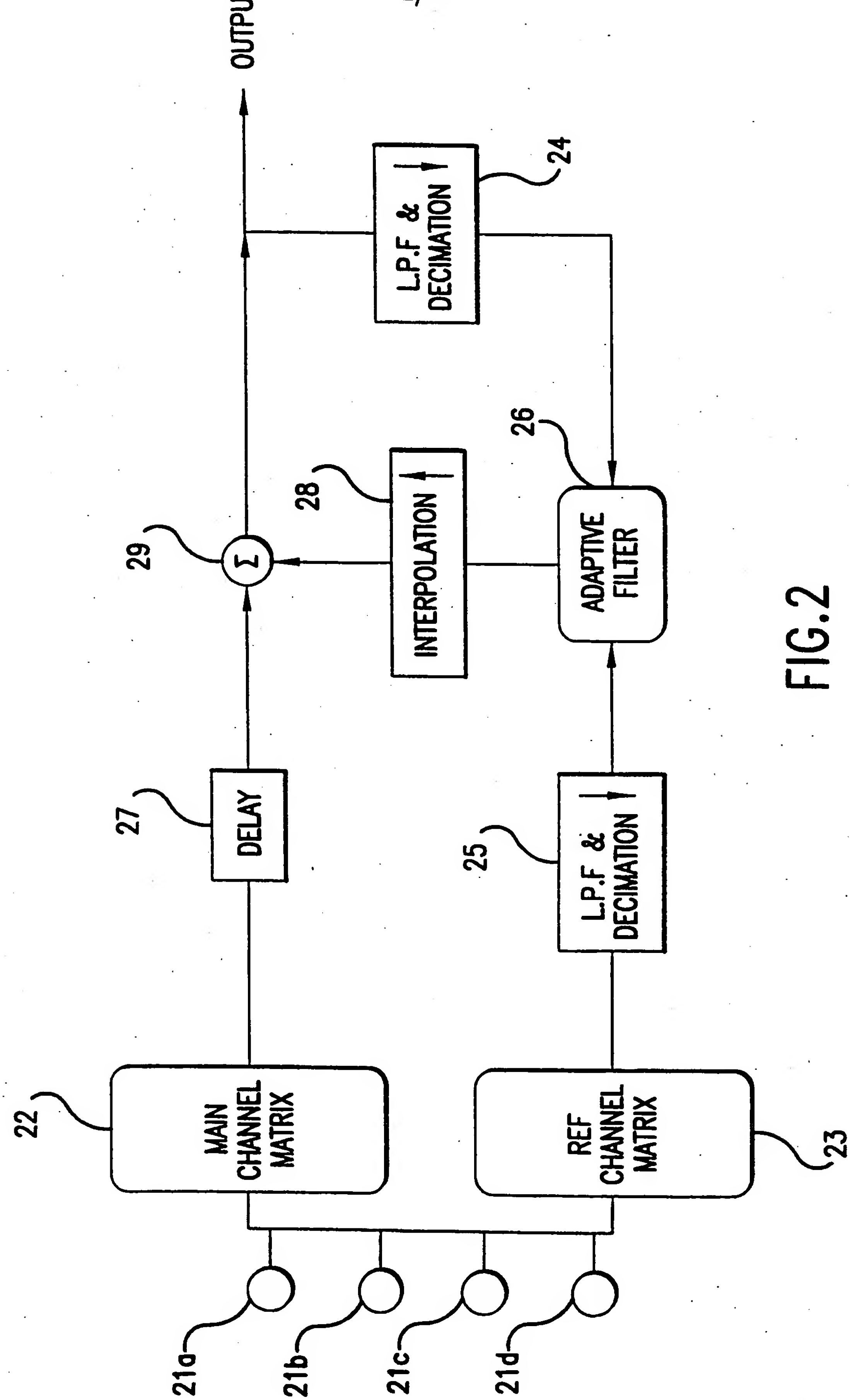
OUTPUT
L.P.F &
DECIMATION
24
INTERPOLATION
28
29
Σ
ADAPTIVE
FILTER
26
DELAY
27
L.P.F &
DECIMATION
25
MAIN
CHANNEL
MATRIX
22
REF
CHANNEL
MATRIX
23
21a
21b
21c
21d

FIG.2

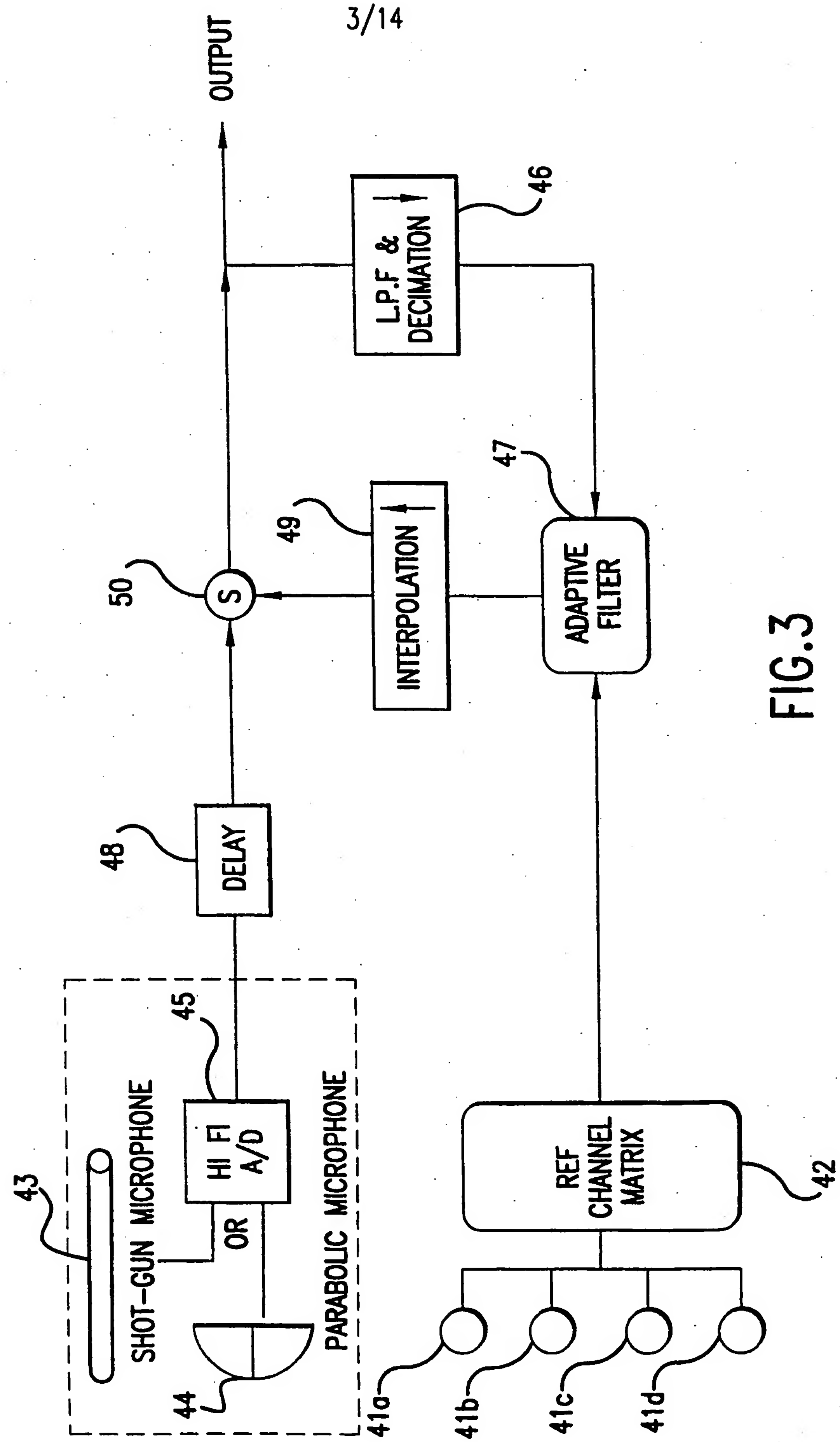
OUTPUT
L.P.F &
DECIMATION
46
47
49
50
S
INTERPOLATION
ADAPTIVE
FILTER
48
DELAY
45
43
SHOT-GUN MICROPHONE
HI FI
A/D
OR
PARABOLIC MICROPHONE
44
REF
CHANNEL
MATRIX
42
41a
41b
41c
41d

FIG.3

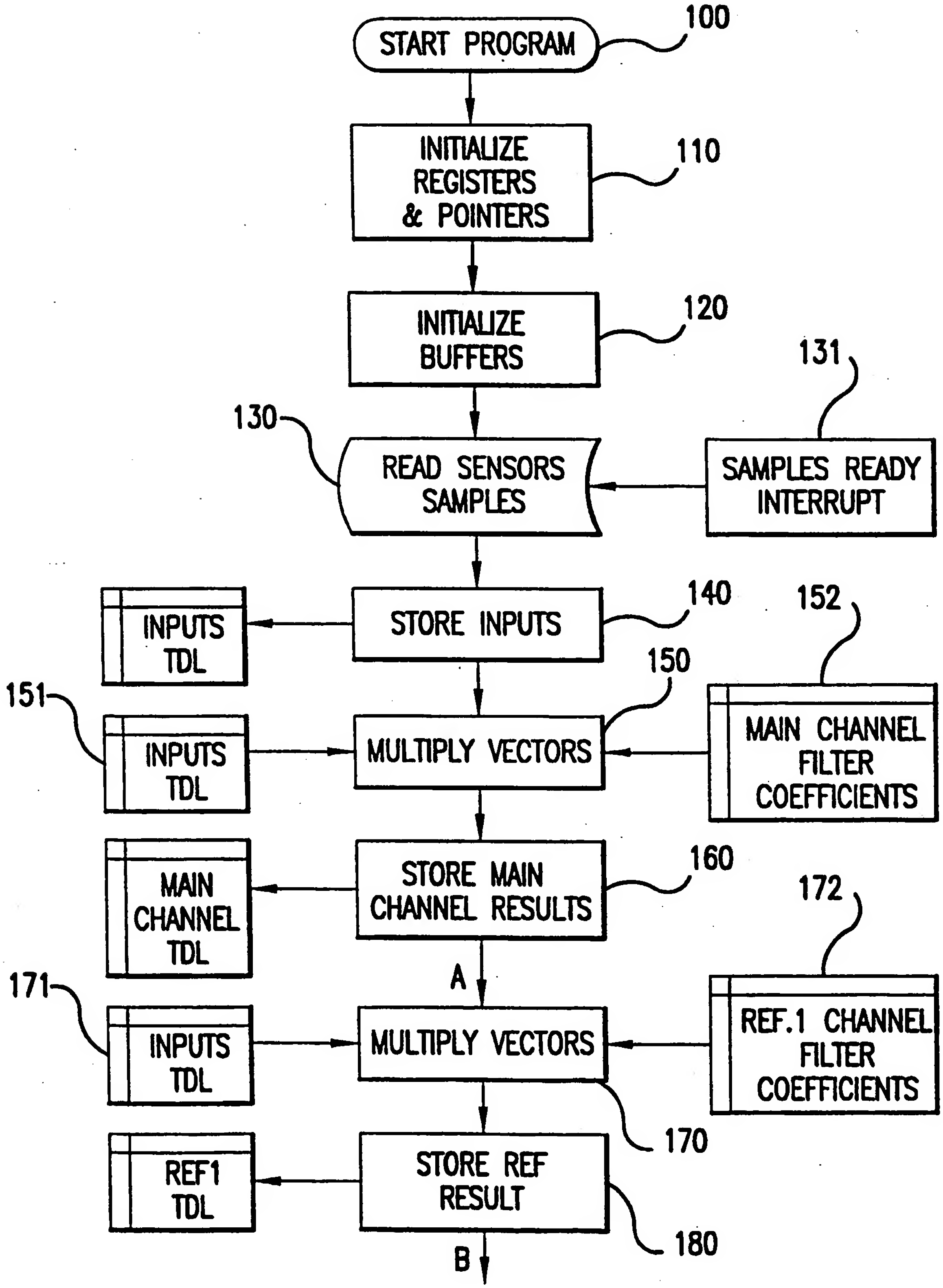
START PROGRAM
100
INITIALIZE REGISTERS & POINTERS
110
INITIALIZE BUFFERS
120
131
130
READ SENSORS SAMPLES
SAMPLES READY INTERRUPT
INPUTS TDL
STORE INPUTS
140
152
150
151
INPUTS TDL
MULTIPLY VECTORS
MAIN CHANNEL FILTER COEFFICIENTS
MAIN CHANNEL TDL
STORE MAIN CHANNEL RESULTS
160
172
A
171
INPUTS TDL
MULTIPLY VECTORS
REF.1 CHANNEL FILTER COEFFICIENTS
170
REF1 TDL
STORE REF RESULT
180
B

FIG.4A

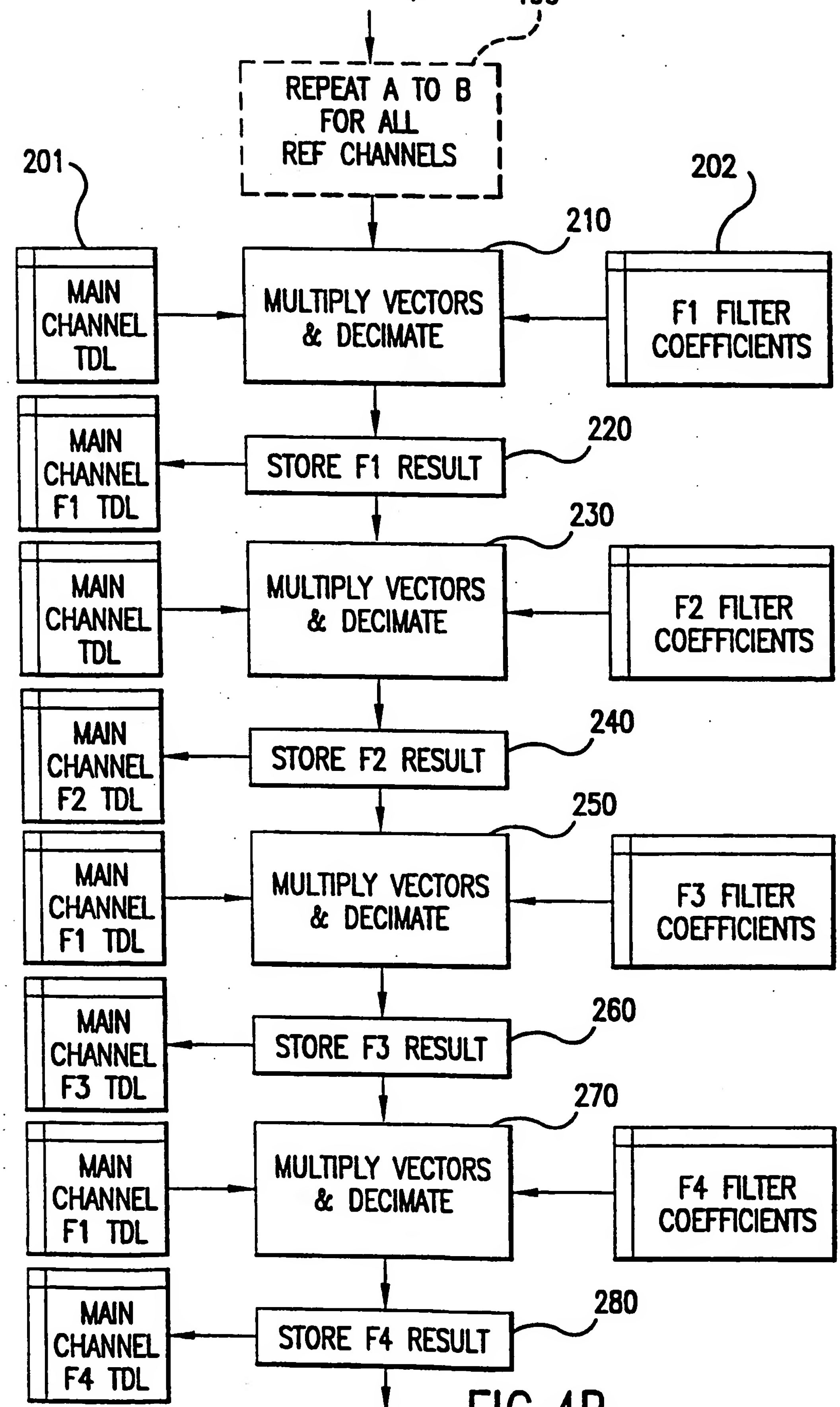
190
REPEAT A TO B FOR ALL REF CHANNELS
201
202
210
MAIN CHANNEL TDL
MULTIPLY VECTORS & DECIMATE
F1 FILTER COEFFICIENTS
220
MAIN CHANNEL F1 TDL
STORE F1 RESULT
230
MAIN CHANNEL TDL
MULTIPLY VECTORS & DECIMATE
F2 FILTER COEFFICIENTS
240
MAIN CHANNEL F2 TDL
STORE F2 RESULT
250
MAIN CHANNEL F1 TDL
MULTIPLY VECTORS & DECIMATE
F3 FILTER COEFFICIENTS
260
MAIN CHANNEL F3 TDL
STORE F3 RESULT
270
MAIN CHANNEL F1 TDL
MULTIPLY VECTORS & DECIMATE
F4 FILTER COEFFICIENTS
280
MAIN CHANNEL F4 TDL
STORE F4 RESULT

FIG.4B

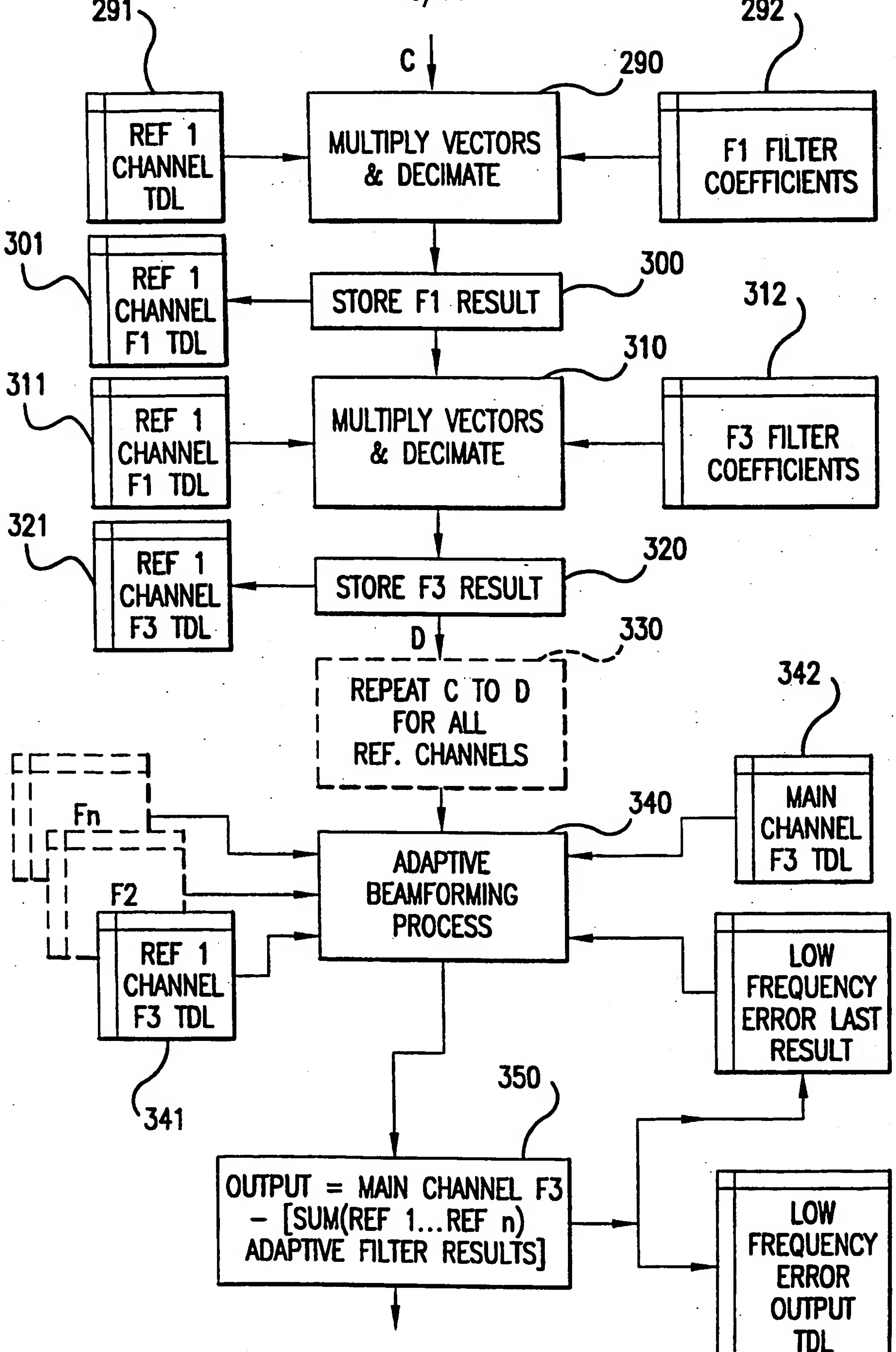
291
C
290
292
REF 1 CHANNEL TDL
MULTIPLY VECTORS & DECIMATE
F1 FILTER COEFFICIENTS
301
300
REF 1 CHANNEL F1 TDL
STORE F1 RESULT
312
310
311
REF 1 CHANNEL F1 TDL
MULTIPLY VECTORS & DECIMATE
F3 FILTER COEFFICIENTS
321
320
REF 1 CHANNEL F3 TDL
STORE F3 RESULT
D
330
REPEAT C TO D FOR ALL REF. CHANNELS
342
Fn
340
MAIN CHANNEL F3 TDL
F2
ADAPTIVE BEAMFORMING PROCESS
REF 1 CHANNEL F3 TDL
LOW FREQUENCY ERROR LAST RESULT
341
350
OUTPUT = MAIN CHANNEL F3 − [SUM(REF 1...REF n) ADAPTIVE FILTER RESULTS]
LOW FREQUENCY ERROR OUTPUT TDL

FIG.4C

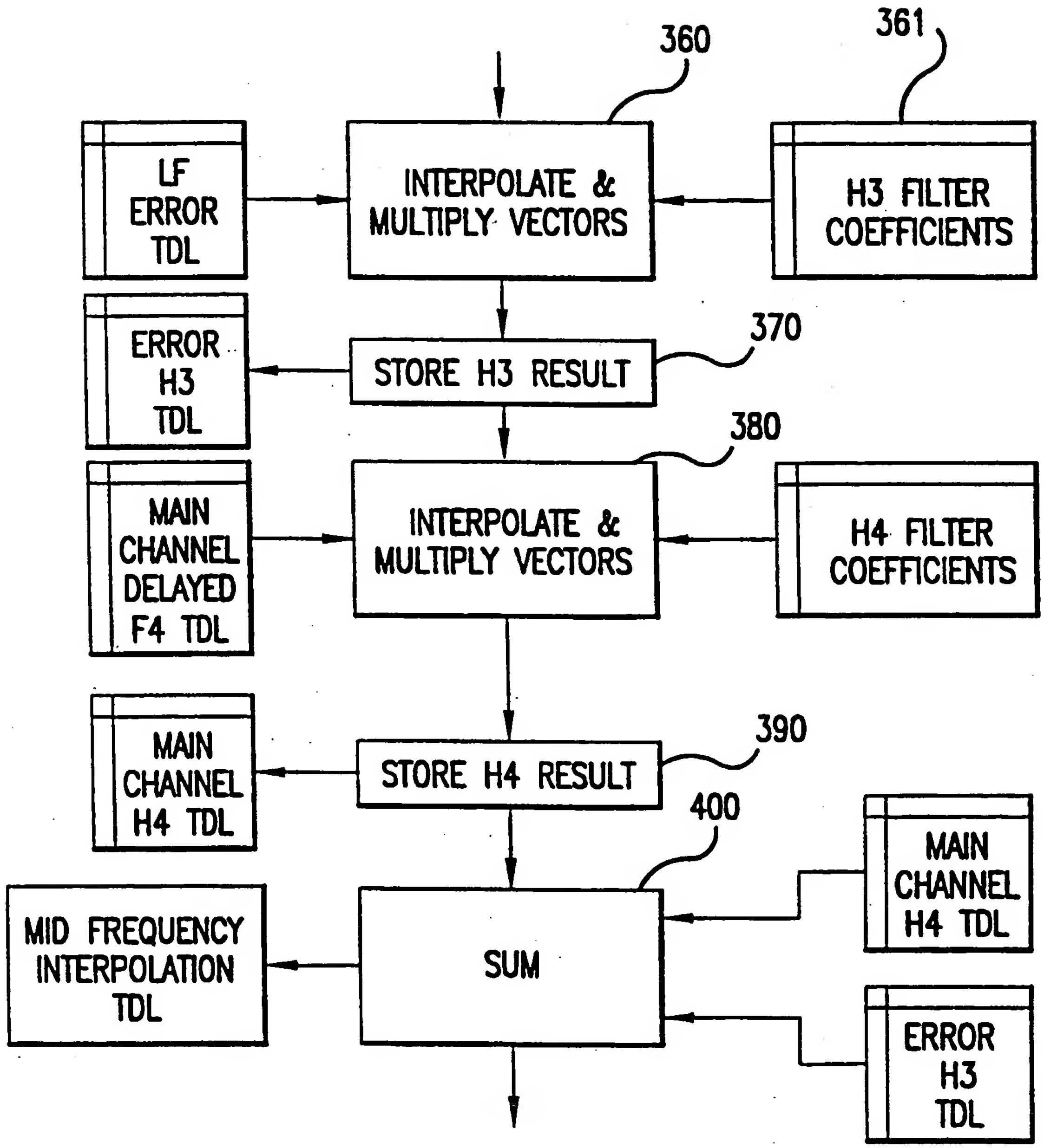
360
361
LF ERROR TDL
INTERPOLATE & MULTIPLY VECTORS
H3 FILTER COEFFICIENTS
ERROR H3 TDL
STORE H3 RESULT
370
380
MAIN CHANNEL DELAYED F4 TDL
INTERPOLATE & MULTIPLY VECTORS
H4 FILTER COEFFICIENTS
MAIN CHANNEL H4 TDL
STORE H4 RESULT
390
400
MID FREQUENCY INTERPOLATION TDL
SUM
MAIN CHANNEL H4 TDL
ERROR H3 TDL

FIG.4D

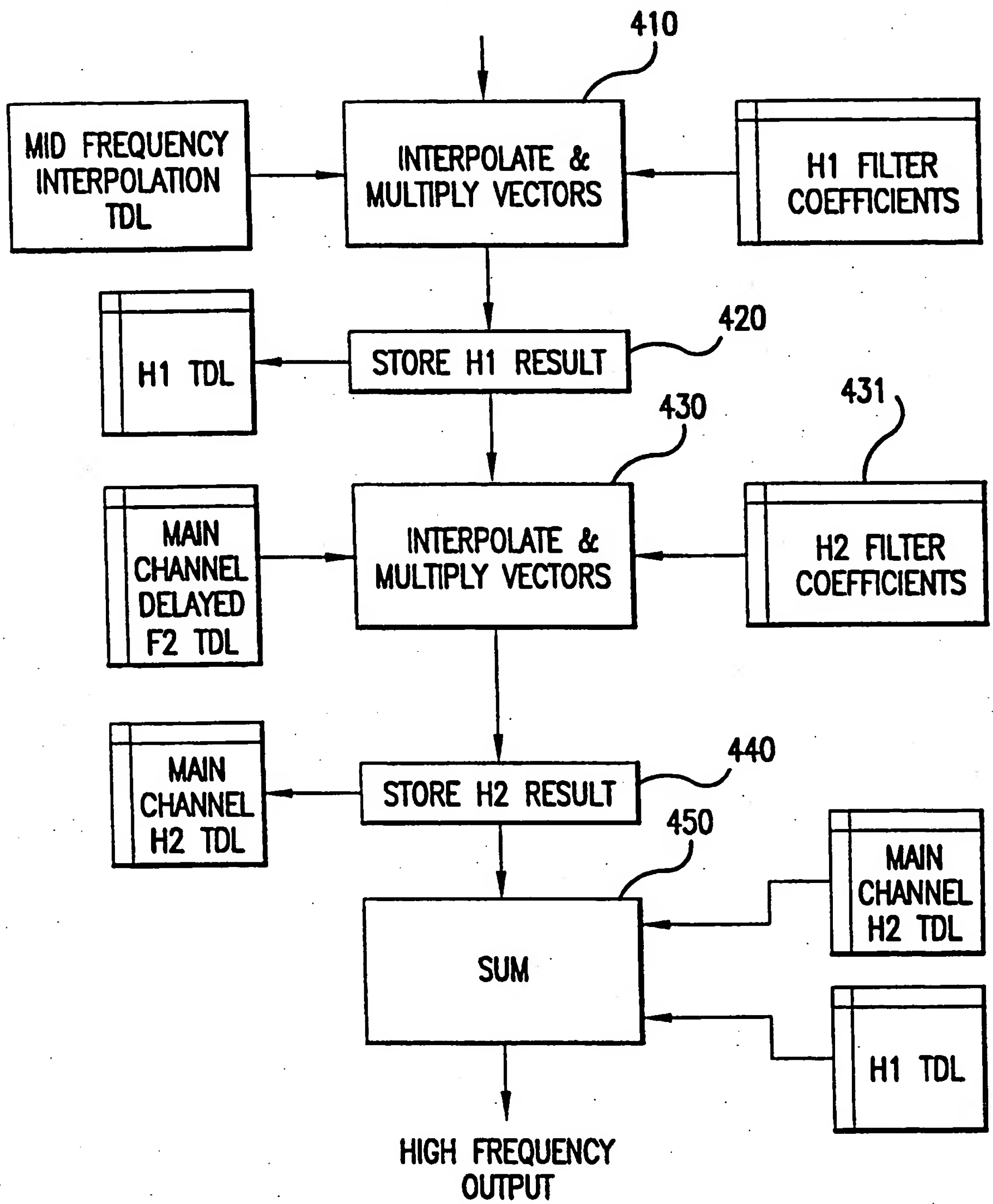
410
MID FREQUENCY INTERPOLATION TDL
INTERPOLATE & MULTIPLY VECTORS
H1 FILTER COEFFICIENTS
420
H1 TDL
STORE H1 RESULT
430
431
MAIN CHANNEL DELAYED F2 TDL
INTERPOLATE & MULTIPLY VECTORS
H2 FILTER COEFFICIENTS
440
MAIN CHANNEL H2 TDL
STORE H2 RESULT
450
MAIN CHANNEL H2 TDL
SUM
H1 TDL
HIGH FREQUENCY OUTPUT

FIG.4E

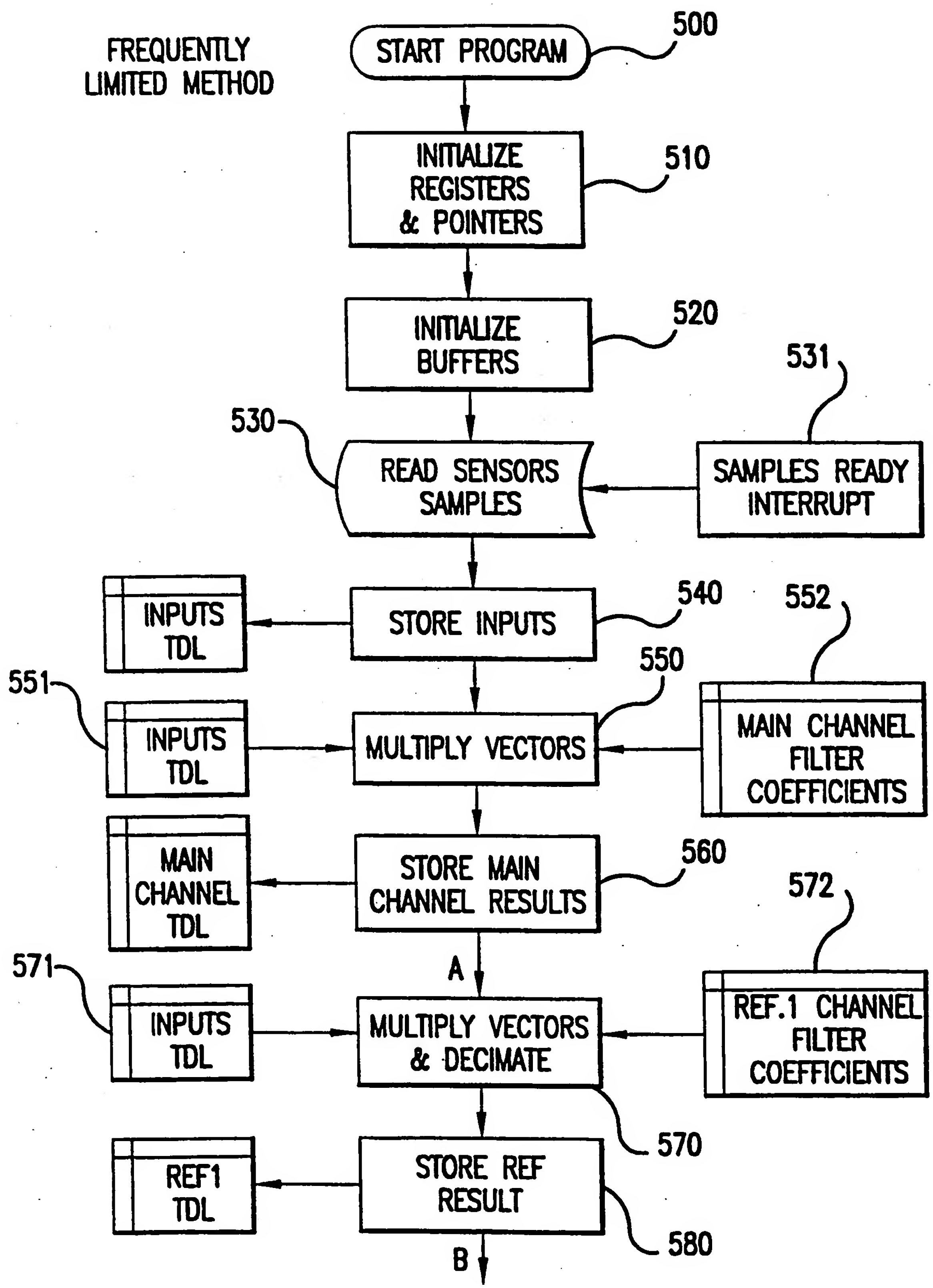
FREQUENTLY LIMITED METHOD
START PROGRAM
500
INITIALIZE REGISTERS & POINTERS
510
INITIALIZE BUFFERS
520
530
READ SENSORS SAMPLES
531
SAMPLES READY INTERRUPT
INPUTS TDL
STORE INPUTS
540
552
550
551
INPUTS TDL
MULTIPLY VECTORS
MAIN CHANNEL FILTER COEFFICIENTS
MAIN CHANNEL TDL
STORE MAIN CHANNEL RESULTS
560
572
571
A
INPUTS TDL
MULTIPLY VECTORS & DECIMATE
REF.1 CHANNEL FILTER COEFFICIENTS
570
REF1 TDL
STORE REF RESULT
580
B

FIG.5A

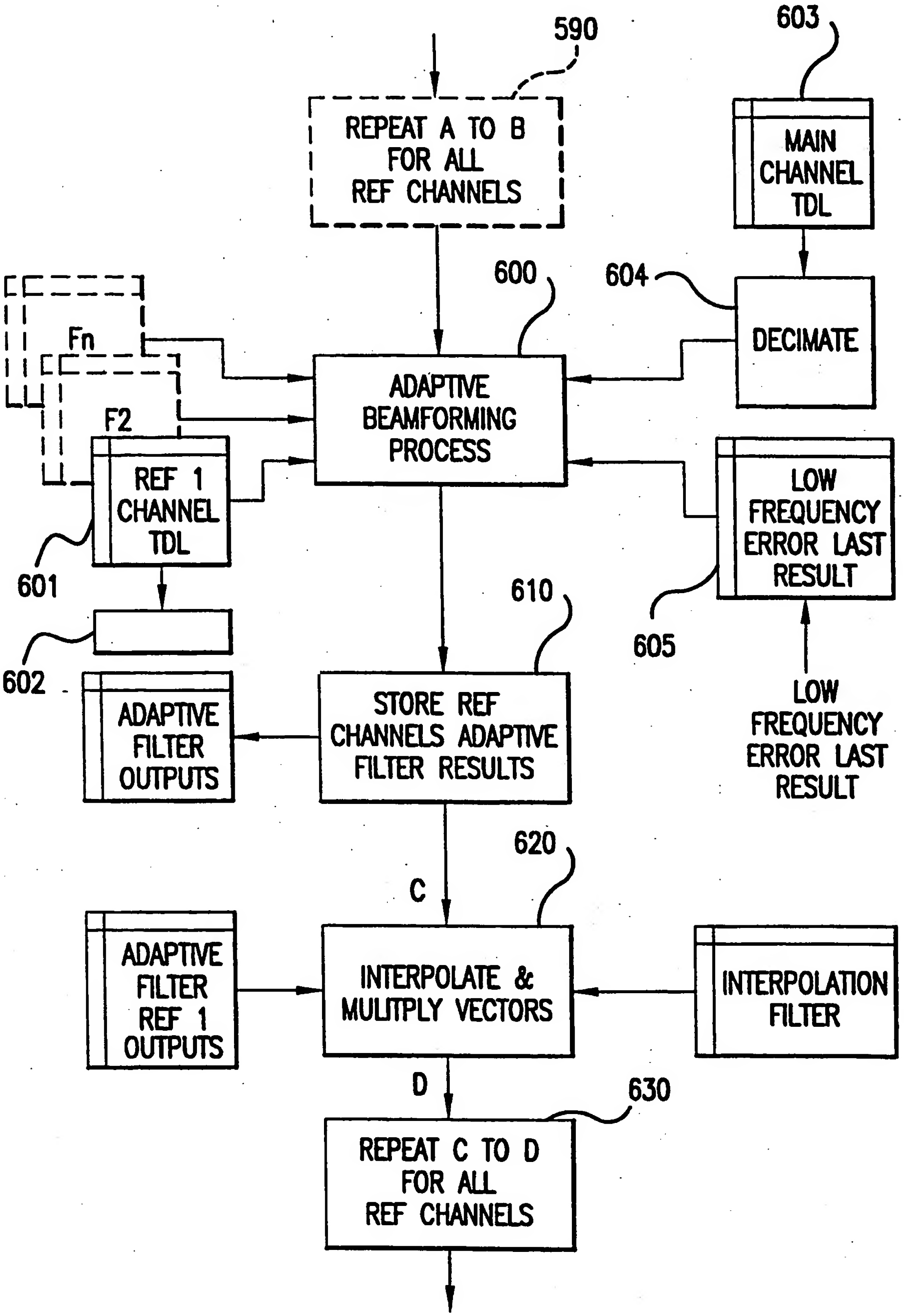
590
REPEAT A TO B
FOR ALL
REF CHANNELS
603
MAIN
CHANNEL
TDL
600
604
DECIMATE
Fn
F2
ADAPTIVE
BEAMFORMING
PROCESS
REF 1
CHANNEL
TDL
LOW
FREQUENCY
ERROR LAST
RESULT
601
610
605
602
ADAPTIVE
FILTER
OUTPUTS
STORE REF
CHANNELS ADAPTIVE
FILTER RESULTS
LOW
FREQUENCY
ERROR LAST
RESULT
620
C
ADAPTIVE
FILTER
REF 1
OUTPUTS
INTERPOLATE &
MULITPLY VECTORS
INTERPOLATION
FILTER
D
630
REPEAT C TO D
FOR ALL
REF CHANNELS

FIG.5B

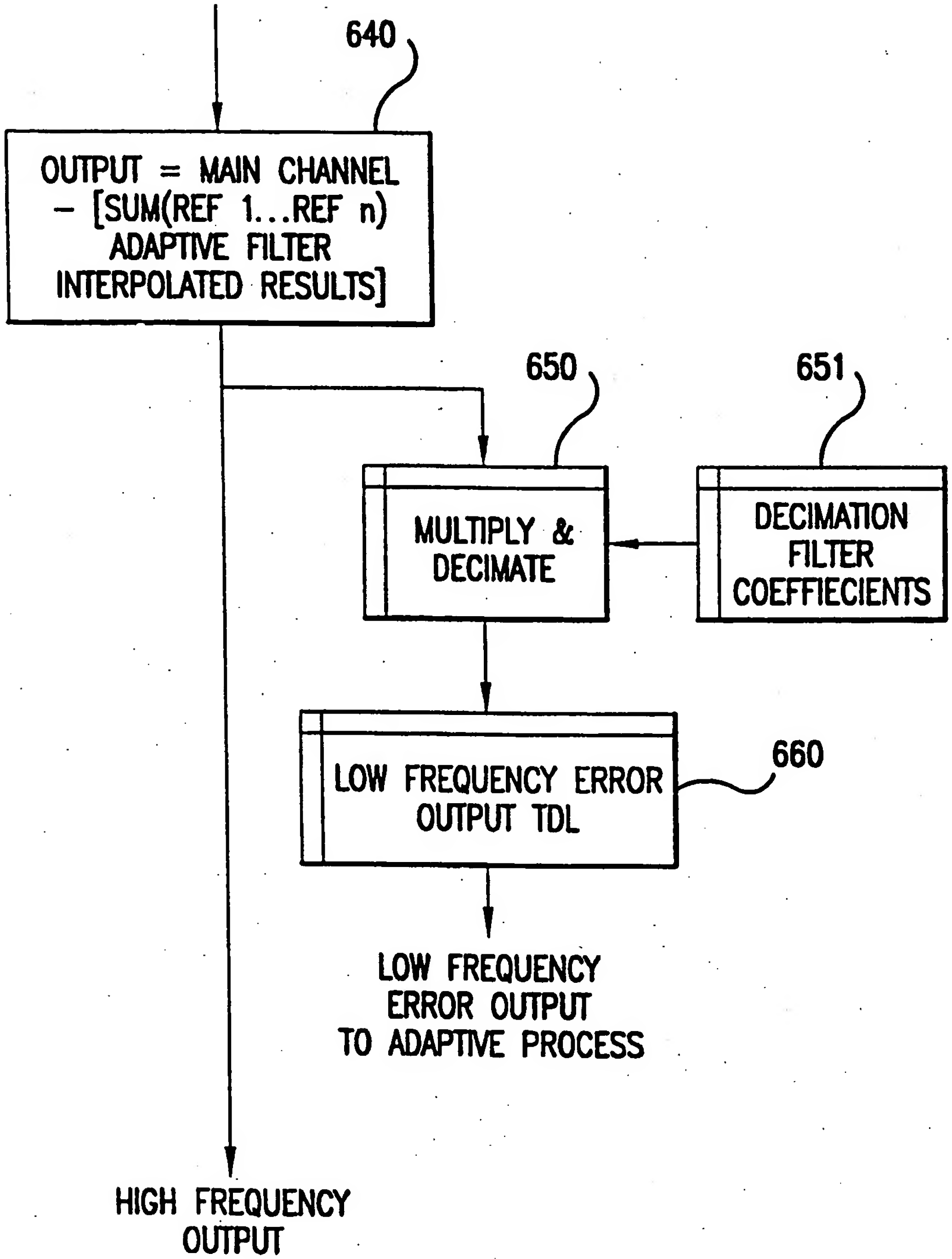
640
OUTPUT = MAIN CHANNEL
– [SUM(REF 1...REF n)
ADAPTIVE FILTER
INTERPOLATED RESULTS]
650
651
MULTIPLY &
DECIMATE
DECIMATION
FILTER
COEFFIECIENTS
LOW FREQUENCY ERROR
OUTPUT TDL
660
LOW FREQUENCY
ERROR OUTPUT
TO ADAPTIVE PROCESS
HIGH FREQUENCY
OUTPUT

FIG.5C

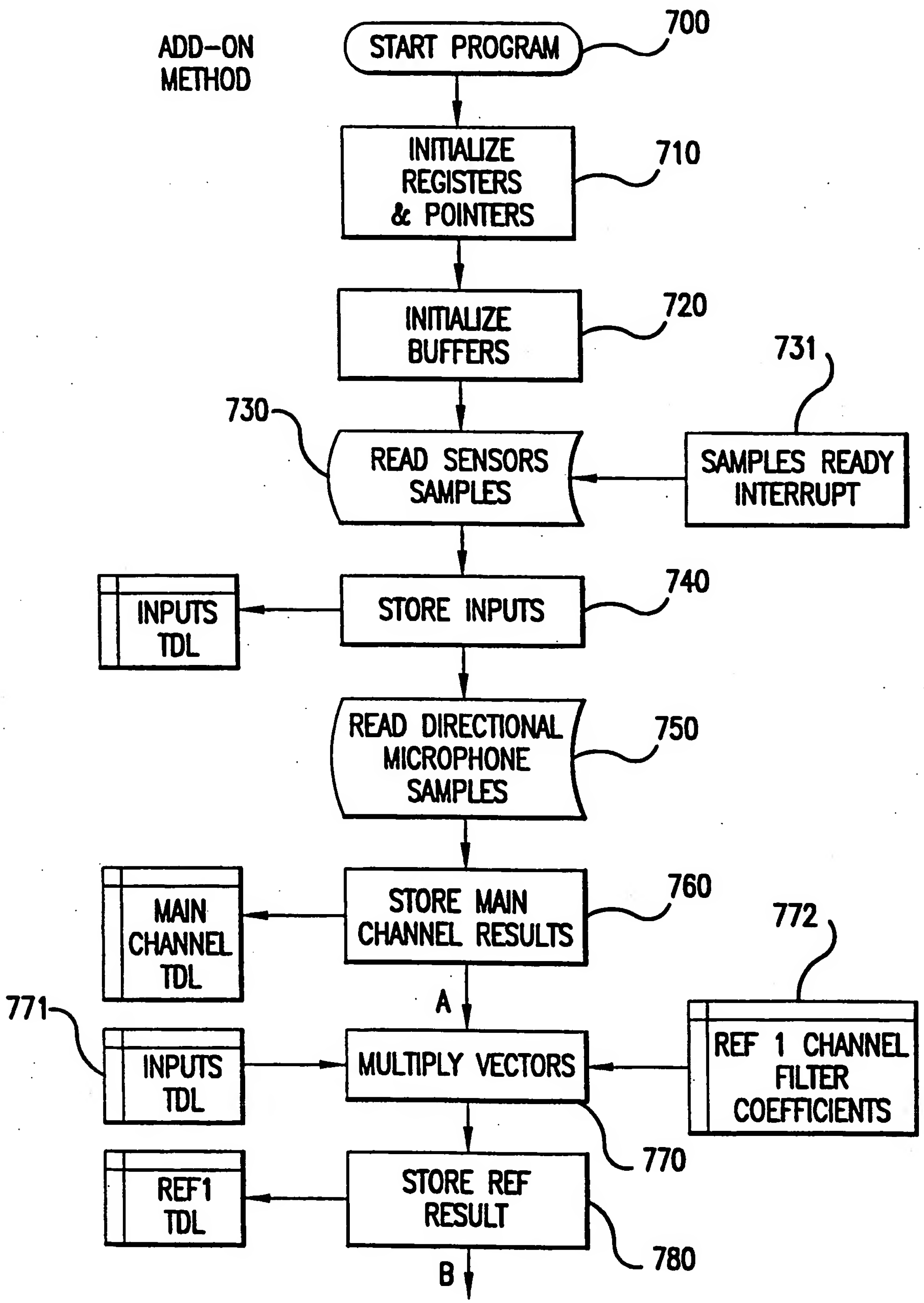
ADD-ON METHOD
START PROGRAM
700
INITIALIZE REGISTERS & POINTERS
710
INITIALIZE BUFFERS
720
731
730
READ SENSORS SAMPLES
SAMPLES READY INTERRUPT
INPUTS TDL
STORE INPUTS
740
READ DIRECTIONAL MICROPHONE SAMPLES
750
MAIN CHANNEL TDL
STORE MAIN CHANNEL RESULTS
760
772
771
A
INPUTS TDL
MULTIPLY VECTORS
REF 1 CHANNEL FILTER COEFFICIENTS
770
REF1 TDL
STORE REF RESULT
780
B

FIG.6A

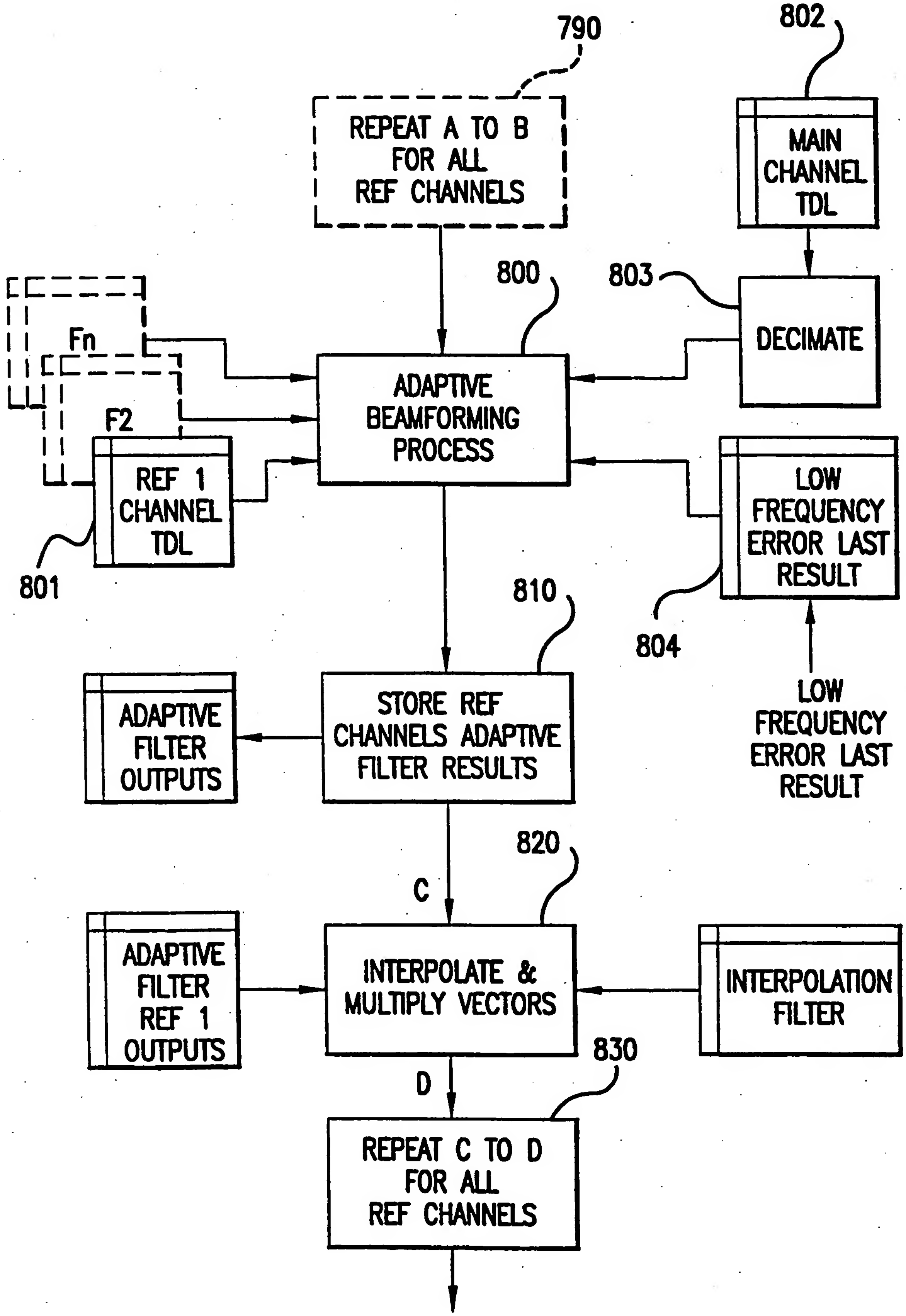
790
REPEAT A TO B FOR ALL REF CHANNELS
802
MAIN CHANNEL TDL
800
803
DECIMATE
Fn
F2
ADAPTIVE BEAMFORMING PROCESS
REF 1 CHANNEL TDL
801
LOW FREQUENCY ERROR LAST RESULT
810
804
ADAPTIVE FILTER OUTPUTS
STORE REF CHANNELS ADAPTIVE FILTER RESULTS
LOW FREQUENCY ERROR LAST RESULT
820
C
ADAPTIVE FILTER REF 1 OUTPUTS
INTERPOLATE & MULTIPLY VECTORS
INTERPOLATION FILTER
830
D
REPEAT C TO D FOR ALL REF CHANNELS

FIG.6B

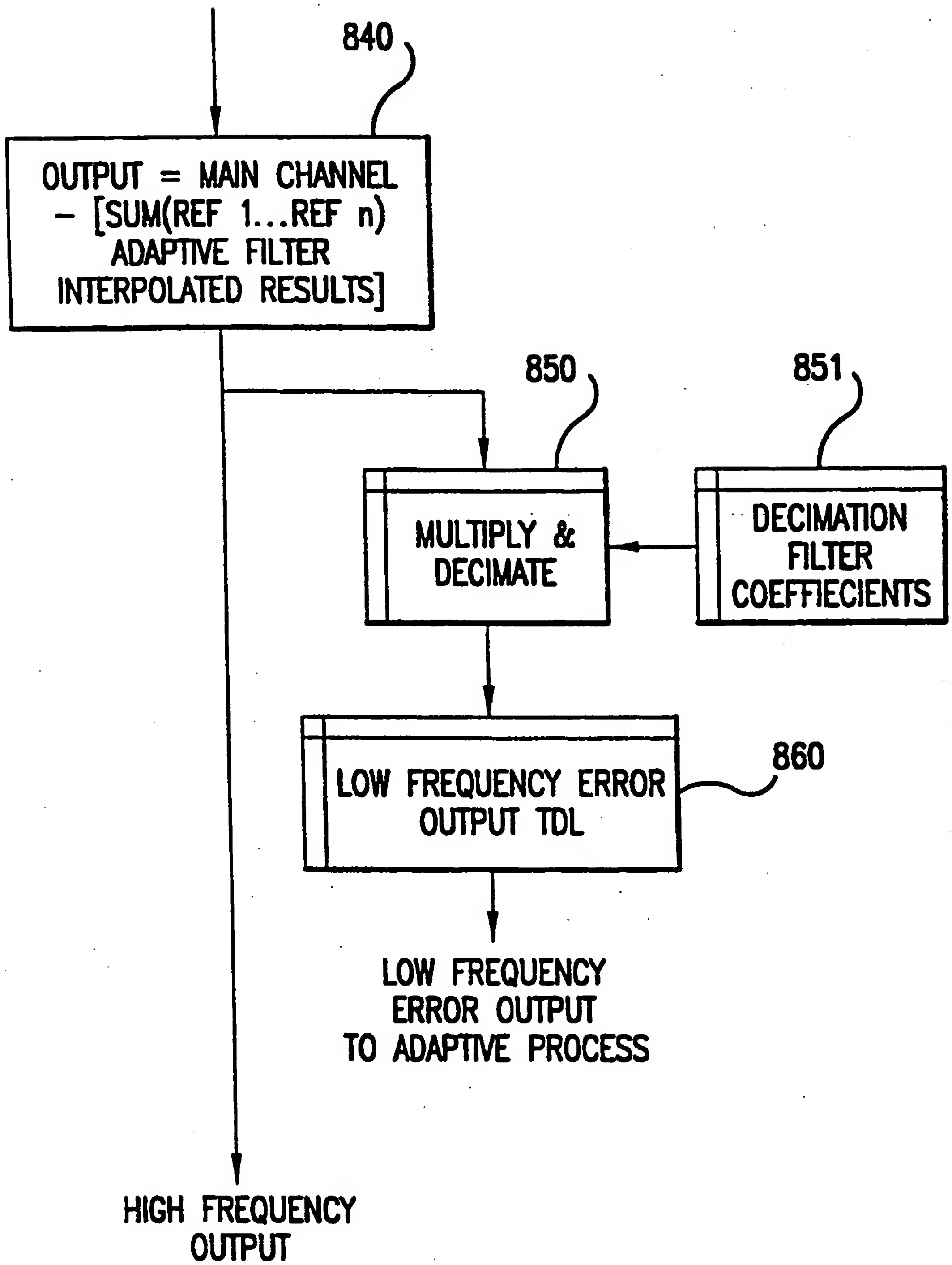
840
OUTPUT = MAIN CHANNEL – [SUM(REF 1...REF n) ADAPTIVE FILTER INTERPOLATED RESULTS]
850
851
MULTIPLY & DECIMATE
DECIMATION FILTER COEFFIECIENTS
860
LOW FREQUENCY ERROR OUTPUT TDL
LOW FREQUENCY ERROR OUTPUT TO ADAPTIVE PROCESS
HIGH FREQUENCY OUTPUT

FIG.6C

INTERNATIONAL SEARCH REPORT

International Application No
PCT/IL 98/00179

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H03H21/00 H03H17/02

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H03H

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
	-/--	

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
16 July 1998	24/07/1998

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Coppieters, C

INTERNATIONAL SEARCH REPORT

International Application No
PCT/IL 98/00179

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	SEBALD D J ET AL: "APPLICATION OF MVDR BEAMFORMING TO REJECT TURBULENCE NOISE IN A DUCT" 1996 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING - PROCEEDINGS. (ICASSP), ATLANTA, MAY 7 - 10, 1996, vol. VOL. 5, no. CONF. 21, 7 May 1996, INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERS, pages 2734-2737, XP000687850	1-3,5
A	see the whole document	6,7,11, 15,16, 20-22, 24,26, 30, 34-36, 40, 48-50, 55,56, 60-62, 66,67
Y A	EP 0 721 251 A (AT & T CORP) 10 July 1996 see page 3, line 15 - page 4, line 22; figure 2	1-3,5 8,12, 17-19, 27,31, 41,45, 51,54, 57,63
A	MUKUND PADMANABHAN ET AL: "RESONATOR-BASED FILTER-BANKS FOR FREQUENCY-DOMAIN APPLICATIONS" IEEE TRANSACTIONS ON CIRCUITS AND SYSTEMS, vol. 38, no. 10, 1 October 1991, pages 1145-1159, XP000278315 see page 1145, column 1, line 18 - page 1146, column 1, line 13 see page 1152, column 2, line 32 - page 1154, column 1, line 11; figures 6,7	1,9,10, 13,14, 28,29, 32,33, 42,43, 46,47, 52,53, 58,59, 64,65
A	EP 0 524 560 A (GENERAL INSTRUMENT CORP.) 27 January 1993 see column 8, line 36 - column 9, line 2; figure 6	74-94
P,L, A	WO 97 50186 A (LAMAR SIGNAL PROCESSING) 31 December 1997 see page 5, line 26 - page 6, line 17 see page 15, line 16 - page 17, line 3; figure 10	74-94

2

1

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/IL 98/00179

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		CA 2073918 A	27-01-1993
		DE 69221970 D	09-10-1997
		DE 69221970 T	26-03-1998
		JP 5226975 A	03-09-1993
		KR 9506765 B	22-06-1995
WO 9750186 A	31-12-1997	AU 3105597 A	14-01-1998